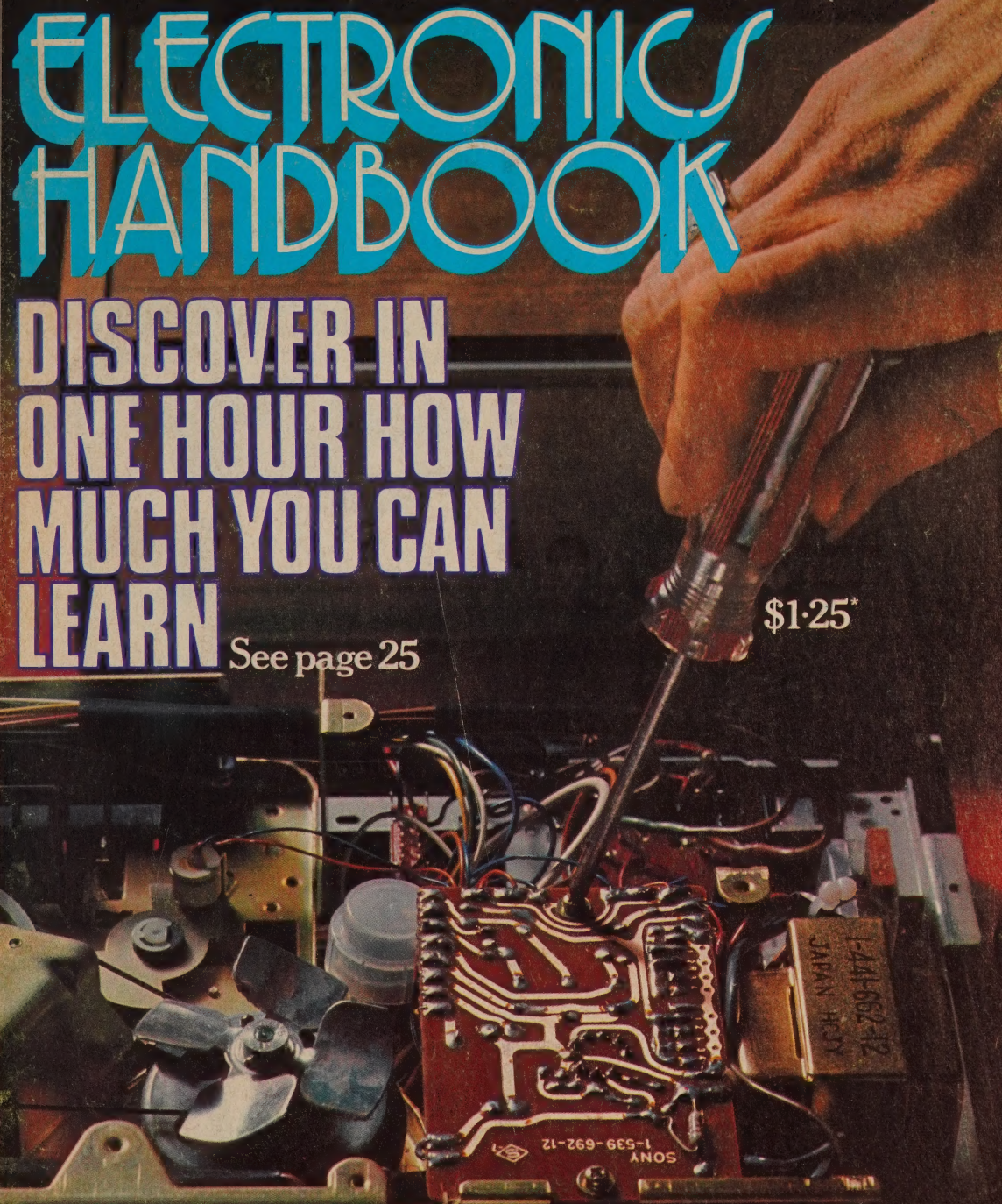
ELECTRONICS HANDBOOK

DISCOVER IN ONE HOUR HOW MUCH YOU CAN LEARN

See page 25

$1.25*



OPEN THE DOOR TO NEW HOBBIES & CAREERS IN ELECTRONICS

Packed with facts on: ✳4-CHANNEL MATRIX & CD-4 SOUND ✳SIMPLE FILTER CIRCUITS ✳SUPERHET CIRCUITS FROM GATSBY'S ERA ✳AMPLIFIER SECRETS YOU SHOULD KNOW ✳CB TRANSCEIVER & ANTENNA THEORY ✳MICROWAVE OVEN THEORY & USE ✳BATTERY APPLICATION & TESTING DATA ✳**PLUS much more for the...** ✳BEGINNER ✳NOVICE ✳HOBBYIST ✳OLD TIMER ✳EXPERIMENTER ✳PROFESSIONAL

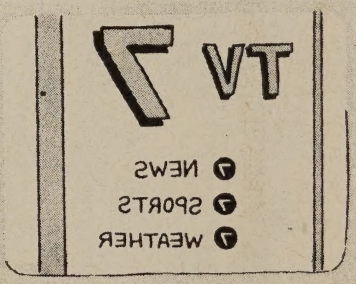
Some Common TV Problems and Their Causes

by Jack Schmidt

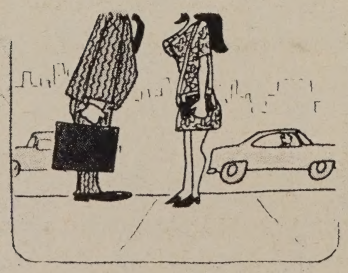
•The kids were playing baseball in the familyroom . . . forget it!

This means you have been watching too long and your dinner is burning.

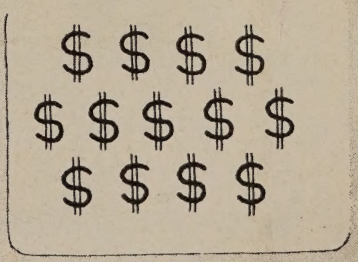
There's nothing wrong with your set! This is an educational channel showing digestion in a rhino's stomach.



Remember the night you made all the adjustments from the back of the set with the aid of a mirror?



Your antenna isn't high enough!



Your set payment is overdue . . .

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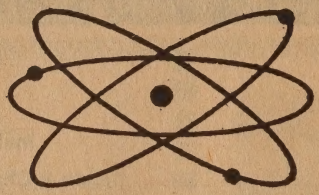
ELECTRONICS HANDBOOK



Inside Matrix Sound

WHEN faced by an inexplicable situation some people are apt to answer that "aerodynamically, the bumble-bee can't fly, but it does." What he means is that *he* does not know a rational reason for an event, but he does know *it can happen.* Fortunately, we in the hi-fi game do not need to fall back on something like "the bumble-bee can't fly." We can give a perfectly logical explanation for anything—if we can't come up with a logical answer, we simply make one up by falling back on the term *psychoacoustics.*

Yes, *psychoacoustics* has carried us safely through many rough seas, such as cassette noise reduction: "If you kill the highs, at low volume you won't really miss them." Fortunately, we usually luck out, as the hi-fi really flies. But unfortunately, *psychoacoustics* did not quite carry us through matrix surround-sound. All the present matrix systems depend to a major extent on what the originators term psychoacoustic effects. With absolutely rotten separation inherent in the 4-2-4 matrices, it was determined that if the listener held his head *rigidly* positioned in the center of the sound field—looking neither to the right nor left but straight ahead—he would hear surround-sound. This is known as the "kitchen-chair-in-the-middle-of-the-living-room syndrome," the *logical* reason for the audiophile's initial lack of enthusiasm for surround-sound, not to forget the distributor's avoidance of 4-channel caused by his customer's lack of enthusiasm.

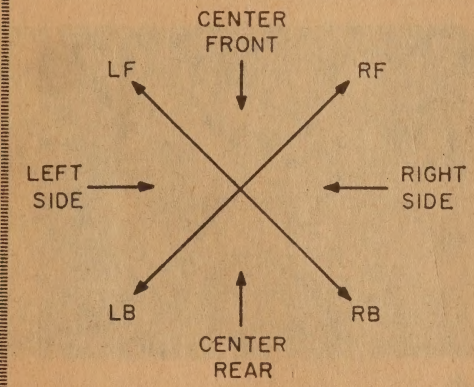
The Big Hang-up. To give you a rough idea of what the potential quadraphile faced, the original SQ papers (theory presentations) "proved" conclusively that the SQ separation was more than adequate because of psychoacoustic effects when the listener faced forward. (And let us not heap all the abuse on SQ, for we use it only as a convenient example; all the matrix systems suffer from the poor-separation-let's-use-psychoacoustics characteristics. The other matrices just push the lack of separation into different quadrants.) Well, if the system was so good, why are we on the third *blend factor* in decoders to produce acceptable separation? Why are the other matrix systems changing the blend, encoding mixing ratios and using rotating mixing angles? The answer is, of course, to improve separation; while psychoacoustics resolves the *kitchen-chair syndrome* it does not provide sufficient quadrant separation for a *walk-through sound field,* one in which a listener in any location will hear a distinctive point-source origination for a specific sound or sounds.

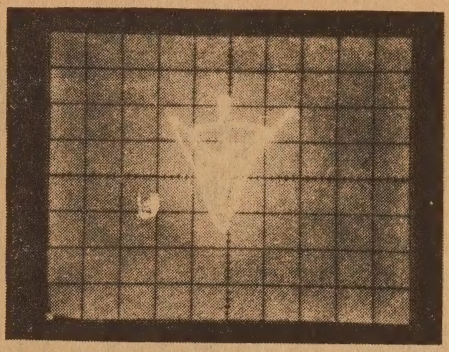
In addition to a high degree of separation, any surround-sound system requires a more or less true playback or decode representation of the original (live) quadrant sound distribution for proper perspective. However, the fact that one might not attain proper perspective does not reduce the possibility of program enjoyment or acceptability, because if the listener has no idea of the original quadrant sound distribution he cannot know something is amiss. Ad-

mittedly, any matrix surround-sound sounds good to the ear; the most we can do is provide additional quadrant enhancement — usually termed separation. In fact, the greater the degree of separation, the greater the listening pleasure—the more spectacular the surround-sound effect—even when matrices are intermixed, such as SQ played through QS or QS through an EVX-44. While quadrant information might shift position due to intermixed matrices, who is to say what is correct? If it's enjoyable, so what!

Getting Apart! Several different methods have been tried to increase separation, as we'll illustrate. Since the SQ system presently has a competitive edge both with manufacturers and consumers, we will concentrate primarily on SQ equipment. The first attempts to increase SQ separation from front to rear involved changing the blend factors, the amount by which one quadrant is blended into another. Most all late-model equipment uses the third and latest blend factor, which provides from 4 to 8 dB separation, depending on who built the decoder. While one might assume all decoders should deliver essentially similar results, such is not the case, as illustrated in Figs. 1 through 4, with multiple exposures of an SQ "star" representing the information in the primary sound field quadrants.

The almost perfect representation in Fig. 1 was provided by the full-logic wavematching SQ decoder in a Lafayette LR-4000B receiver. (We'll get to logic circuits later.) The loops indicate approximately 5% phase shift some-





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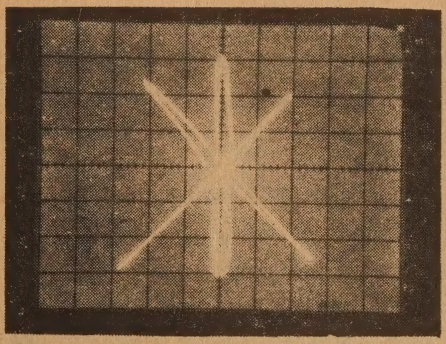

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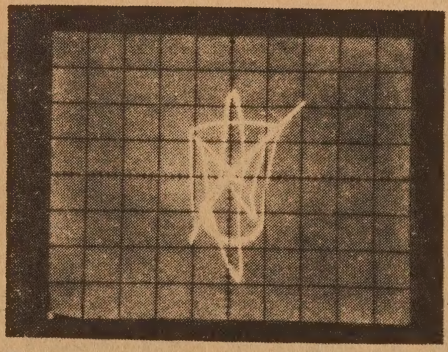

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Each photograph represents the sound field that would be established by a quadraphonic system. The upper left is left-front (LF), the upper right is right-front (RF), the lower left is left-back (LB), the lower right is right-back (RB). The center of the picture—actually the center of an oscilloscope display—is the sound center mid-point between all outputs. Center-front information is created when the LF and RF signals are equal and in-phase. If the signals are in-phase but unequal in amplitude, the centered signal moves towards the stronger of the two sides. The same is true of center-rear signals developed between LB and RB. Information shown at the extreme sides between LF and LB and RF and RB are developed when information in the two paired corners (such as LF and LB) are in phase—as occurs with front and rear signals. No information is displayed between the two associated corners (paired corners) when the corner signals are out of phase.

The length or excursion of the oscilloscope display indicates the relative sound level from the quadrant. The absence of a display in a particu-

(Continued on next page)

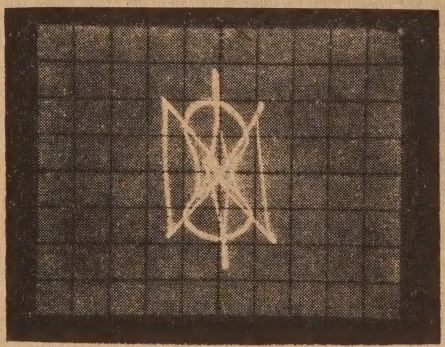
where in the system; either in the pickup or amplifier—a typical low degree of expected phase shift. Note that side information—quadrant information appearing in other quadrants—is almost negligible, represented by the small blur in the center of the sound field (center screen).

Fig. 2 is the same SQ star processed through a popular well-known decoder. Very large degrees of side information as well as poor rear-quadrant performance are indicated.

Fig. 3: another leading SQ decoder. Results are sharply improved (remember, the oscilloscope scale is linear, not logarithmic, in dBs). In fact, Fig. 3 is typical of most SQ decoders.

Fig. 4 is thrown in just to shake you up. Study it closely before reading further; try to figure out what it represents. . . . Fig. 4 is the SQ star through a Pioneer amplifier with a "regular matrix" (assumed SQ) decoder. The so-called butterfly pattern, often confused with proper decoder operation, represents in-phase LF/LB and RF/RB. Center-front and center-back are correct. The hour-glass pattern which is in phase front and rear is supposed to be the RB signal. Sound scrambled? You betcha. Sound good? You betcha. So let's use this opportunity to digress.

The Picture of Music. Fig. 5 shows 500 milliseconds of program, a vocalist with band recorded in QS and played back through a QS decoder. Note the predominant front-centered information (the singer) with the same information in the center-rear at reduced volume.



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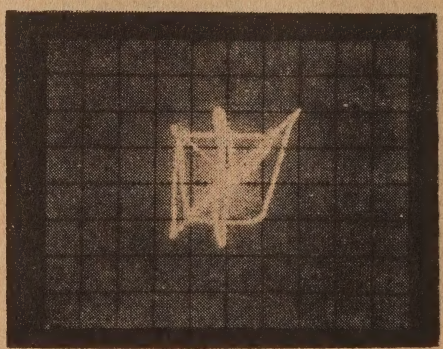

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lar corner quadrant does not indicate no output; rather, it indicates relatively lower output in relation to the other, and in particular its associated, corner.

Illustrations with a gray background were made using sinewave input signals; those with a black background are program material: (1) The SQ "star" pattern obtained through a full-logic decoder shows essentially no side signal spill into other quadrants. The overall effect is almost equal to discrete surround-sound. (2) The SQ star processed through a popular basic SQ decoder results in increased side information and rear output not up to that obtained from the front channels. (3) The SQ star through a more typical basic SQ decoder indicates quite good all-around performance. The in-phase side spill in all quadrants is typical. (4) The SQ star when processed through a basic QS decoder shows substantial reorientation of the original quadrant information. The hourglass pattern indicating an in-phase front and rear signal originally was the left-back. But it all comes out sounding good to the ear. (5) 500 milliseconds

Fig. 6 is the same 500 msec of program, only this time played through an SQ decoder. The vocalist has shifted predominantly right center, with considerable side information at the RB. Yet even with the quadrant shifts the over-all sound is good to the ear.

It's obvious, from the SQ star and program waveforms shown in Figs. 1 through 6, that the basic matrix separation is far from outstanding; some might say it's also far from good. Unfortunately, any matrix has just so much inherent separation because the available frequency range determines the *total* separation for all quadrants. The separation can be stretched from side to side with reduced center-front to center-back separation, or center-front to center-back can be stretched with a resulting loss in side-to-side separation. The total available separation can be pushed, squeezed or stretched, but always at a reduction in separation somewhere else.

First attempts to improve on the inherent SQ separation involved new blend factors, which were explained back at the beginning of this article. The second serious modification to the SQ system was the so-called *logic decoder.*

Enter Logic. The logic decoder was (and is) a teeter-totter (seasaw) gain-riding amplifier. When the decoder senses predominantly front information, the overall gain is shifted to the front, creating greater front-to-rear separation. Depending on the decoder, the same gain-riding can be applied to rear information. There are a few side effects other than the teeter-totter

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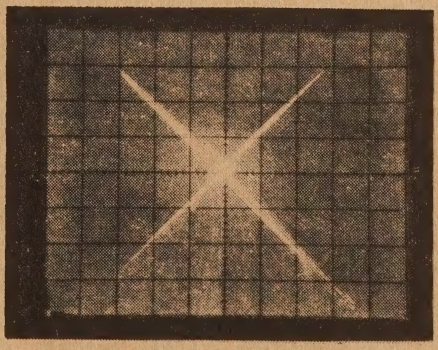

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of program material recorded in QS and decoded in QS shows the vocalist at center-front with some separation to the rear. (6) When the QS-recorded vocalist is decoded in SQ, his voice moves toward the center right with increased rear output. (7) SQ star decoded by a logic decoder (compare with 2 and 3) shows increased center-front to center-rear separation, though there is reduced separation at the extreme sides. (8) 500 milliseconds of a band with predominant trumpets at the LF. (9) When the 500-millisecond record of a band is processed through a full-logic decoder, the LF information—the trumpets—is enhanced, creating an almost discrete-appearing sound output. (10) When the QS-recorded vocalist shown in Figs. 5 and 6 is processed through a full-logic SQ decoder, the logic circuits "see" the vocalist at the right center and provides right-side enhancement, moving the vocalist to a decided right location. (11) Even when all four corner signals are sim-

(Continued on next page)

gain-riding which we won't get into because they don't seriously affect the total enhancement, and the whole idea is passé anyway.

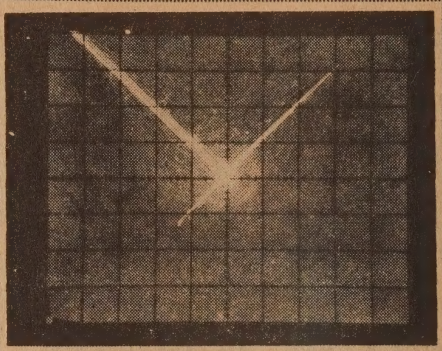
Fig. 7 illustrates the effect of *logic* separation enhancement. Compare Fig. 7 with the basic decoder output shown in Fig. 3; note how much of the "garbage" has been cleaned up by front-to-back logic. The center-front and the center-back signals are particularly clean, with sharply reduced opposite-quadrant spill. Note, however, the increase in the in-phase left and right side signals, a trade-off for increased center-front to center-rear separation.

The ultimate—at least for the present—SQ decoder is the so-called full-logic, which includes wave-matching. In the full-logic decoder, both the center and corner signals are processed for optimum enhancement. Gain-riding is used for the center front and rear signals; a signal that appears primarily in the center-front causes the decoder's gain to shift sharply towards the front, producing a decided forward or center sound. The same effect takes place for center-rear signals. The wavematching handles the corners; signals with predominant corner information similarly shift overall gain toward its corner. The end result of the gain-riding and wave-matching is the SQ star shown way back in Fig. 1, the output of the Lafayette LR-4000B. With center front and rear balance set for equal output levels, corner output is very close, and side-effect spill is negligible. Other than the minor phase shifts represented by the loops in each leg of the star, the decoding is essentially the ultimate that can be expected of a matrix system limited to a 20 to 20,000-Hz bandwidth. Also, accept that other enhanced matrix systems can accomplish the same ends; unfortunately, they take more hardware than the SQ system and are not presently available.

Of course, there are people who have the same attitude towards tests run with sinewave signals as those expressed towards psychoacoustics at the beginning of this article. To them, the total sound quality as it affects *program* is the most important consideration. Figs. 8 and 9 represent 500 msec of band music; the trumpets at LF are predominant in the original scoring. Fig. 8 is the program processed by a basic SQ decoder; Fig. 9 is the same 500 msec of program processed by a full-logic decoder. Note how the full-logic enhances the LF trumpets, reducing the spill into the opposite quadrants. The listening effect is almost that of discrete surround-sound.

While we're still on the subject of full-logic let's take a look at what it does to programs recorded with the same QS matrix. Fig. 10 is the same 500 msec of program used to illustrate Figs. 5 and 6. If you recall, the vocalist on the QS recording moved towards the right and rear when processed by an SQ decoder. Now, as shown in Fig. 10, full-logic seizes the SQ's predominantly right-side vocalist and moves him even further to the right with even greater signal strength in the RB. In fact, through speakers the vocalist definitely appears at RB (right-back)—and it's a better, more spectacular effect than originally recorded. (Of course, if we had an enhancement QS decode the program would sound even more spectacular—but there presently is no such thing available.)

Finally, let's end by showing what happens when all four corners receive a simultaneous in-phase SQ signal, as this will show all the "garbage" produced by side signals. Fig. 11 is the full-logic decoder; the output is very good, with the total side effects represented by the small smear at the center. Fig. 12 is the output from a basic SQ decoder produced by one of the most famous names in hi-fi—we will save them embarrassment. Actually, Fig. 12 has considerable rear-quadrant information, but the levels and phasing are so badly scrambled they



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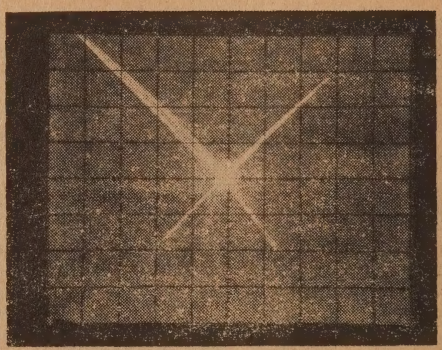

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ultaneously applied, the full-logic decoder has very little side signals generated—those forming the small smear at the center. (12) When all four corner signals are applied to a popular basic SQ decoder, the result is obviously not what was intended. The sound output is notably front oriented, particularly to the left-front. (13) Simultaneous application of the four corner SQ signals through a *universal* decoder shows quite good results.

cancel through the oscilloscope equipment, whose display happens to match what a listener will hear if located at the center of the sound field. Fig. 13 is the SQ output from an EVX-44 decoder. Results are quite good; the *hash* at the right (which might not show in the magazine reproduction) is in-phase information between RF and RB; this degree of in-phase information is not evident from LF to LB. However, as with all things about matrix sound, it sounds very good to the ear.

It's important to take note of the fact that except for laboratory test records which provided the sinewave test signals, and the laboratory prepared 500-msec program tapes recorded on a phase-matched Revox A77 tape recorder, all playback equipment were standard quality items as might be used or purchased by the average stereophile, for the purpose of the tests were simply to show what surround-sound characteristics will be realized from hi-performance consumer equipments. The basic equipment consisted of an Empire Troubador turntable, Empire 1000ZE/X and Shure V15 Type III pickups (and only those results are shown which could be duplicated on both pickups), a Lafayette LR-4000 receiver and the latest Tektronix oscilloscope equipment specially modified for 4-channel display.

If we were forced to select one single outstanding aspect of the matrix situation, it would be that *all* decoders should have full wave-matching logic or any electrical system that could provide essentially the same degree of quadrant enhancement. It is a shame that any matrix systems were released before their overall performance reasonably approached that of full-logic, for not only does full-logic greatly enhance the performance level of its own encoding; it also provides spectacular enhancement and effects from other encoding formulas. And that's the logic of it all. ■

Inside 4-Channel CD-4 Sound

☐ The time has come to stop talking of cabbages and kings, matrix and decoders, SQ and QS, ambience and psychoacoustics. Discrete 4-channel sound is here, both in equipment and records, and it's totally compatible with your present stereo system.

The discrete 4-channel system finding its way into our homes is tabbed CD-4—Compatible, Discrete 4-channel. Now, you can play and thrill to the most realistic sounds on records. You can feel that elusive "sound field" that brings the fuller dimension to music as experienced in live performances. CD-4 captures and reproduces the natural reverberation, presence, movement and resolution heard by the conductor, musicians and audiences that have never been fully realized on disc recordings before! CD-4 is the culmination of musical excitement.

Unfortunately, for the reading public, too much has been said about CD-4 before it was fully understood. Many editors, reporters and authors promulgated myths mixed with facts until the CD-4 picture clouded in confusion. So, let's set the record straight now! Forget whatever you learned about CD-4. Let's pick up the threads of facts and weave a new picture—honest, reliable, and—by all means—*correct*.

The CD-4 technique of discrete 4-channel sound maintains four separate channels of sound from the recording studio all the way through the audio reproduction system to the four loudspeakers. Unlike matrix systems, the four discrete channels are never blended, encoded or scrambled into two channels and then decoded. Consequently, there is no blending or overlapping of sound from channel to channel. Instruments intended to be heard through only one speaker in the CD-4 system are *only* heard through that speaker, and *not* the others. There is no psychoacoustical crutch to lean on, no complex logic circuits to ride gain, no tricks, no fast talk to confuse you.

CD-4 requires some special equipment. If you want to go discrete 4-channel today by adding on to your present stereo system, here is what you need:

● **A high performance record player.** But, you have one in your stereo system. If it's of the quality of the Benjamin/Lenco L75, Elac/Miracord 650, Dual 1229, Empire 598 Mark II, Garrard Zero-100, VC VL-8, PE 3012, Pioneer PL-61, or Thorens TD-125AB Mk II, among others, you are all set. If not, this stereo system component needed upgrading before you started this article!

● **An extended range phono pickup with Shibata stylus.** The ordinary elliptical stylus in the best stereo pickup is not quite suitable for playing CD-4 records. So far, only Audio Technicia has a line of CD-4 phono pickups with Shibata styli, and JVC offers their 4MD-20X model. One important technical feature of these phono pickups is extended frequency range—required for CD-4, a bonus feature when playing back ordinary stereo discs.

● **A CD-4 disc demodulator.** Some of the units currently available are JVC's 4DD-5 and Panasonic's SE-405H. Also, JVC offers a combination automatic changer and CD-4 demodulator, the 4VC-5244. Several manufacturers offer similar products.

● **A 4-channel amplifier.** If you are adding on to a stereo system, you'll be in the market for a piggyback amplifier—a component that provides two additional power amplifying stages for the rear channels. A reasonably low-cost unit is the

Lafayette LA-524. JVC's VN-5101 should be considered, too. At best, the piggyback is a stop-gap idea that can be avoided by planning to switch to a 4-channel amplifier unit. Your best bet will be to invest in a new 4-channel amplifier.

● **A 4-channel receiver.** Stereo receivers require a piggyback amplifier for 4-channel discrete operation exactly the same as for stereo amplifiers. However, this is the one component area where considerable consideration should be given to replacing the stereo receiver with a *quality* 4-channel unit.

● **Two more speakers.** The most reasonable selection of rear speakers would be identical units to match the front pair. Forget about the cheaper, or poorer performance speakers for use as rear speakers in a 4-channel system. This practice was practical for ambience and early matrix decoders, but not so for CD-4. Each channel must be matched to the other three for CD-4 is high-fidelity throughout!

If you have a matrix or discrete 4-channel system that plays SQ records or 4-channel, 8-track tape, you can swing to the CD-4 system immediately with JVC's 4DD-5 demodulator and a Shibata stylus pickup.

Once you have a CD-4 system, you'll need CD-4 discs to head discrete programming. RCA is already producing discrete 4-channel records called QUADRADISCS and the selection of records, currently at 50 albums, is being rapidly expanded to include all kinds of music, with 20 new releases each month. Other companies, such as Warner, have been licensed to make CD-4 records, and their discrete CD-4 discs may be in the record shops by the time you read this.

Disc Price Is Compatible. Unlike many matrix records sold at an increase of one dollar or more, CD-4 records sell for the same suggested retail prices as ordinary 2-channel stereo records. Since CD-4 records are compatible, record stores don't have to carry a double inventory and pass a price increase on to the consumer. And the myth about CD-4 discs playing only half the time of a standard stereo LP (because they carry twice as much information) isn't so. CD-4 record now have essentially the same playing time as 2-channel discs.

While you shouldn't use your present stereo phono pickup with a CD-4 demodulator (a CD-4 cartridge with Shibata stylus must be used in higher-quality light-tracking tone arms), you can use the CD-4 cartridge on your old stereo records. There is a Panasonic demodulator, the SE-405C, that includes a cartridge with a conical stylus which, in less sophisticated playback equipment, can track CD-4 records at up to 3.5 grams. The results are *good* to *great*! The CD-4 phono pickups often enhance stereo playback because of their excellent, extended, flat frequency response. Also, the Shibata stylus —shaped differently than the standard elliptical stylus—rides on a different portion of the groove wall. Hence it will often deliver excellent sound from standard stereo records with grooves worn from numerous plays by a standard stereo stylus.

Now, here are some facts learned from actually playing, experimenting and enjoying CD-4 recordings.

Extended repeated play of CD-4 discs by the Shibata stylus does not seriously degrade the high frequency subcarrier information contained in the grooves. Earlier forecasts of 5, 10, or a maximum of 20 plays on a CD-4 disc have been proven false. Four-channel playback quality lasts nominally as long as stereo playback quality does; thus, CD-4 playback is good for the life of an average stereo disc.

Keep It Clean. CD-4 records are very susceptible to the effects of dust and dirt in the grooves. If the sound gets "hairy" wash the record in room-temperature water. If a mild detergent is used, rinse thoroughly under tap water. Do not wipe dry, but allow the records to drip dry. Then play the record through once and the stylus will remove the dust from the grooves. Be sure to clean the stylus tip with a stylus brush. (The stylus tip always must be kept scrupulously clean.) A "dustbug" or similar record cleaner tracking ahead of the stylus is a must-have accessory for CD-4 records.

One important point should be mentioned. Many audio reporters and authors claimed that the bottom-end frequency limit of the CD-4 record was near 100 cycles. This is definitely not so; as any listening test will indicate CD-4 discs are capable of going down to the deep bass. The myth began when the press was told that CD-4 disc masters were cut at approximately one-third the normal playback speed. Since the typical stereo recording goes down to near 40 Hertz, the press wrongly concluded that the CD-4 recording must be limited about 100 Hz minimum. This is not so. (Normal stereo cutting techniques can go as low as 10 cycles but no one can hear it, most phono pickups cannot pick it up, and many record manufacturers cut off almost everything below 50 Hertz anyway.) Current recording practice brings the frequency range down to 40 Hertz and below when needed. CD-4 recordings match stereo recording playback frequency range.

Though the best Shibata-equipped phono-pickups track at forces about 1 to 1½ grams above that of the best stereo pickups, the Shibata stylus contacts a larger area of the groove. Thus, the actual pressure of the stylus on a unit area of the groove wall is not higher than an elliptical.

To Keep The Subcarrier . . . CD-4 pick-ups must be used with low-capacity cables supplied with the demodulator. The turntable's standard shielded output cables cannot be used because they attenuate the high-frequency subcarrier output from the CD-4 phono pickup. Most better turntables have standard phone jacks con-

How CD-4 Gives You Discrete Sound

☐ The biggest design problem facing CD-4 was how to put four discrete channels of sound into a V-shaped record groove. This seemingly insurmountable technical problem was solved by modulating two additional audio channels into the groove in much the same way as is done in FM stereo broadcasting. See Fig. 1.

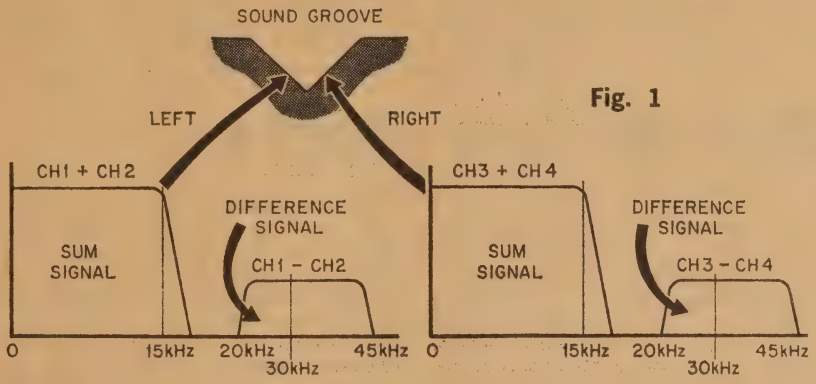


Fig. 1

In CD-4 records, each side of the groove is at a 45-degree angle to the surface of the record as it is in stereo discs. See Fig. 2. While the conventional 2-channel stereo record contains one channel of sound on each side of the groove, the CD-4 disc has two channels of sound on each side of the groove. The sum of the left-front and left-rear channels (#1 and #2) is recorded by the conventional method on the left-channel side of the record groove, and the sum of the right-front and right-rear channels (#3 and #4) is recorded on the right channel side. (See Fig. 3 for speaker placement.) Thus, CD-4 is totally compatible with the present stereo system when played back by a stereo cartridge. No loss of frequency response or undesired mixing of front channels occurs. When using CD-4 for 4-channel discrete playback, the high frequency difference signal on each groove side is used to demodulate the front from rear signal, restoring them to their original status as discrete signals, unblended and in full fidelity. All this was made possible on a CD-4 recording by employing new techniques in the recording process. By low speed cutting, new cutting stylus, and an automatic noise reduction system, to name a few, CD-4 playback from a disc is possible in your home today!

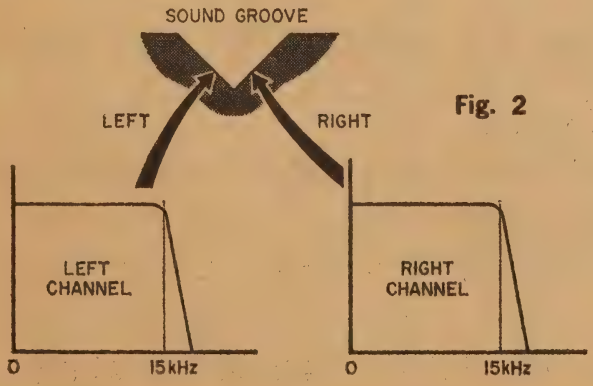
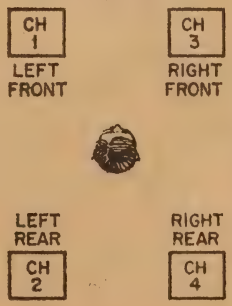


Fig. 2

Fig. 3



cealed under the turntable. Simply unplug the standard shielded cables and substitute the low capacity cables. Empire turntables have a special connecting jack with attached cables for which an adapter plug for low capacity cables is available from CD-4 equipment manufacturers. Try not to use cable extensions. If you must, be sure to use only low capacity cables.

What Does The Future Hold For CD-4? More and more equipment manufacturers will switch to CD-4 as they have done in the Orient and Europe. A holding down of record costs will spirit buying by the consumer, producing a bonus crop of new CD-4 recordings as the market volume blossoms. Even CD-4 FM broadcasts are in the works. Extensive successful field tests and actual broadcasts in San Francisco and Toronto have proven CD-4 broadcasts to be technically feasible. The FCC is currently studying proposals for quadrasonic FM broadcasting. The future is bright for CD-4, the *compatible, discrete 4-channel* audio system. ■

Filter Design–Part 1

□ Filters are perhaps the most widespread type of circuit used in electronics. The tuning circuits which every TV and radio set uses to select the desired station are filters; the IF circuits which shape the bandpass of TV sets are also filters, as are the standard roll-off circuits in tape decks and phonograph amplifiers. FM receivers use filters to remove the pre-emphasis put into the signal at the transmitter, while yet another type of filter is used at the transmitter to provide the pre-emphasis in the first place. And, of course, all electronic circuits drawing power from the AC power line use a filter to smooth the ripple from the pulsating DC delivered by the rectifier.

Although they are found everywhere, filters have long been something of a mystery to the experimenter. For example, if an experimenter needs to design a filter for his circuit and turns to a textbook for assistance, he will be met with a barrage of *poles, zeroes, s-planes, integro-differential equations,* and a host of other subjects, all of which presuppose that he has at least a graduate degree in electrical engineering. So, although these texts certainly contain the information for designing excellent filters, the techniques are so well hidden under the advanced math that the average experimenter can't make use of them.

However, there are some very simple techniques for filter design which may be easily used by anyone with a very elementary math background to make filters usually more than adequate for the job at hand. In the following pages we shall outline some of these simple approaches to filter design, which should enable you to "roll-your-own" filters as the project requires.

What Is a Filter? A filter is an electronic frequency-sieve. Just as a sieve can hold back large particles in a mixture and pass small ones, a filter can eliminate the low frequencies in a signal and pass the high frequencies. Such a filter is called a *high-pass filter.* Placed at the antenna terminals of a TV set, a high-pass filter can eliminate the unwanted low-frequency interference from nearby amateur or CB transmitters, and pass the wanted, higher-frequency TV signals.

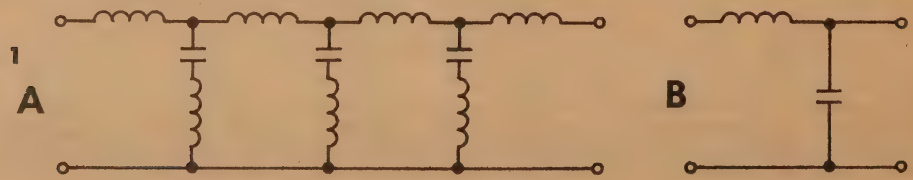
Unlike a normal mechanical sieve, however, a filter can also be built to pass "large particles" (low frequencies) and eliminate "small particles" (high frequencies). Such a filter is called a *low-pass filter,* and can be used, for example, to eliminate the high-frequency scratches, pops, and hisses from a record reproducing system while passing the music signals through to the loudspeaker.

A third type of filter is an even more clever "sieve," for it eliminates both very large *and* very small particles, (low *and* high frequencies), passing only particles of a specific size (signals of a particular frequency or group of frequencies). This type is called a *band-pass* filter, for it passes only a narrow band of frequencies, eliminating signals both above and below the desired signal frequency. Every radio and TV receiver uses a tunable bandpass filter at its input, to select signals from the desired station while eliminating all signals on higher and lower channels.

The fourth and final type of filter is the opposite of a band-pass filter, because it eliminates a particular frequency, while passing all others. This is called *band-elimination filter,* or, if the band eliminated is very narrow, it is called a notch filter.

Complex or Simple. Although there are

FIG 1



only four basic types of filters —high-pass, low-pass, band-pass, and band-eliminate—the circuits that can perform these functions are many and varied. They can be even more complex than the low-pass filter shown in Fig. 1A or they can be as simple as the two-element low-pass filter shown in Fig. 1B.

Measuring a Simple Filter. To understand what a filter is trying to do, let us make a laboratory set-up to measure the performance of the very simple filter of Fig. 1B, using the test set-up of Fig. 2. Here, a variable-frequency oscillator is connected so as to produce any desired audio frequency at the input terminal of the filter, while an audio frequency voltmeter measures the amount of signal emerging from the filter. Starting out with the oscillator set at some very low frequency, such as 10 Hz, we find that the filter has very little effect, passing almost the entire oscillator signal through to the output meter. This you might expect, since the capacitor is almost an open circuit at low frequencies, and the filter will therefore act almost like a series resistor alone, which, in this circuit, has negligible effect on the signal.

As the frequency is increased, however, the capacitor becomes a lower and lower impedance, effectively shorting out the signal as it emerges from the filter. The degree of "shorting out" depends upon the frequency; at very high frequencies almost no signal emerges from the filter.

The clearest display of the behavior of a filter is made by graphing the results obtained in the above experiment. If we plot the meter reading versus the audio oscillator's frequency, we obtain the graph of Fig. 3.

As you can see, the filter is removing the higher frequencies, as shown by the downward sweep of the graph as the frequency is increased. The graph is called the *frequency response* of the filter.

Enter the Decibel. Although the plot of Fig. 3 is easily made and understood, it is not the most useful way to display filter output versus frequency, and hence is not the way frequency response is normally graphed. Usually, instead of reading voltage off the voltmeter's scale, its decibel scale will be read by the experimenter (every good audio-freqency meter has a decibel scale in addition to its normal voltage scale), and decibels will be used on the graph instead of volts. Also, instead of marking off the frequency scale of the graph in equal steps of 1,000 Hz, 2,000 Hz, 3,000 Hz, etc., we mark it in equal steps of 1,000 Hz, 10,000 Hz, 100,000 Hz, etc. When these changes are made in the graph, a *very* useful display results, as shown in Fig. 4.

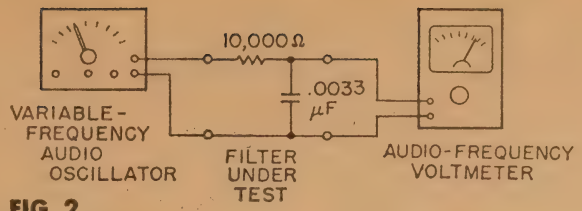


FIG 2

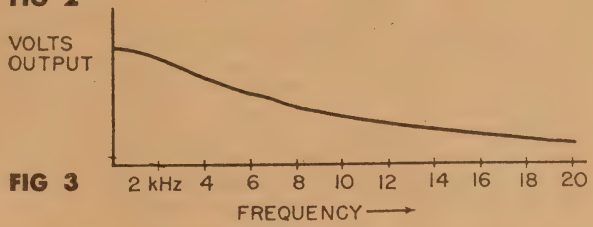


FIG 3

What makes Fig. 4 so useful is that it is essentially two straight lines connected by a short curved line, as shown in Fig. 5 and for purposes of practical design, *the frequency response may be approximated by two straight lines* as shown in Fig. 6.

Note that in Fig. 6, a corner is formed where the two straight lines intersect. A perpendicular line, dropped straight down from this corner, hits the frequency scale at a point known as the *corner frequency*. This is a very important frequency, and is generally taken as the cut-off frequency; that is, the frequency above which the filter rejects signals. The corner frequency

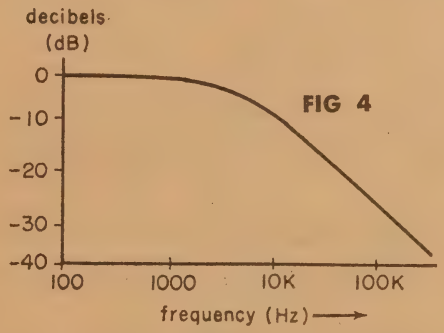


FIG 4

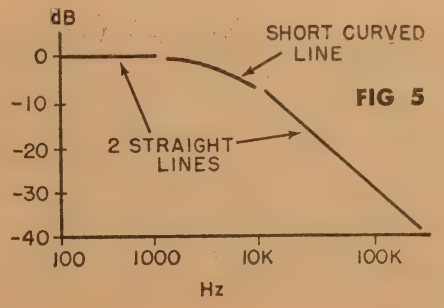


FIG 5

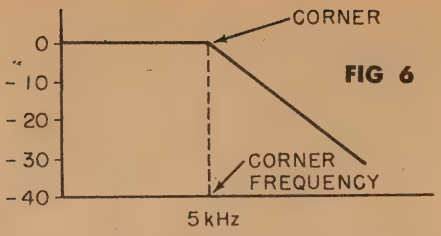


FIG 6

for the particular filter shown is 5 kHz. Frequencies lower than 5 kHz are passed well by the filter; frequencies above 5 kHz are passed relatively poorly.

Rules for Cornering. To make the straight-line approximation of frequency response for a given resistor/capacitor low-pass filter such as the one measured above, you need know only two facts:

1. The corner occurs at a frequency given by

$$f_c = \frac{159{,}000}{RC}$$

. . . where R is in ohms, and C is in microfarads.

2. The downward-sloping straight line falls off at such a rate that between any two decades (such as between 10 kHz and 100 kHz, or between 250 kHz and 2,500 kHz) the line will drop by 20 decibels.

Using these two facts, let us draw the approximate frequency response of a filter with R=2,000 ohms and C=0.22 microfarads.

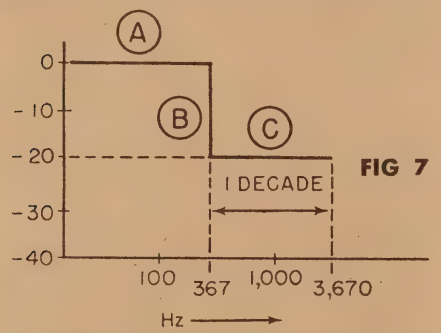


FIG 7

First the corner is calculated as

$$f_c = \frac{159{,}000}{RC} = \frac{159{,}000}{2{,}000 \times .22} = 367 \text{ Hz}$$

As shown in Fig. 7, draw a horizontal line *A* from the zero dB mark out to 367 Hz, then a line *B* down 20 dB, and then a horizontal line *C* over a distance of one decade, from 367 Hz to 3,670 Hz. Now draw a straight line *D* from the point where *A* and *B* intersect, through the tip of *C*, as shown in Fig. 8. You have now sketched a very good approximation to the frequency response of the RC low-pass filter.

Down to Earth. In a real-life situation, a filter doesn't lie on a lab bench, driven from an audio oscillator and having its output measured by a voltmeter, but instead resides in a circuit, where it is driven by a very business-like signal source with a job to do, and delivers its output to a loudspeaker, headphones, telephone line, TV picture tube, or what-have-you. In such a circumstance, our simple RC filter likes to be driven from a low-impedance source, such as a 6-ohm speaker output, and likes to deliver its output to a high impedance such as the base of a transistor amplifier or a pair of high-impedance headphones. Ideally, the high impedance load should be 100 times larger than the low-impedance driving source, although you can get away with as little as a 10 to 1 ratio. At this ratio, however, performance can not be accurately predicted by the simple relationships given in this article.

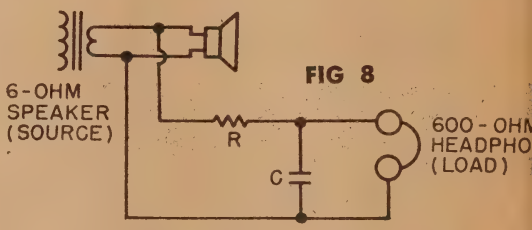


FIG 8

For a good example of a practical use, consider the case of an SWL who would like to drive a 600-ohm headset from his 6-ohm speaker output, and wants to filter the output to eliminate the frequencies above 3,000 Hz. The circuit he would use is shown in Fig. 9.

He must first calculate the value of R for his filter, using the formula

$$R = \sqrt{R_s R_L}$$

where R_s is the source impedance, 6 ohms and R_L is the load impedance, 600 ohms By the formula,

$$R = \sqrt{6 \times 600} = \sqrt{3600} = 60 \text{ ohms}$$

(Use the EIA value, 62 ohms)

He must then calculate the capacity, C, through an algebraic rearrangement of the equation given earlier for corner frequency:

$$C = \frac{159{,}000}{f_c R}$$

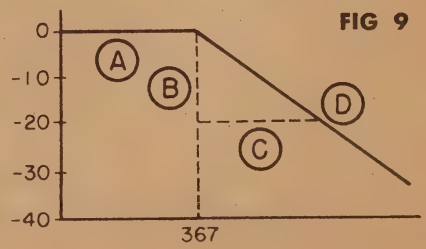


FIG 9

where f_c is the desired corner frequency,

3,000 Hz

and R is the resistance above, 62 ohms Therefore,

$$C = \frac{159{,}000}{3{,}000 \times 62} = .855\ \mu F$$

Replacing the 0.855 *uF* value with the nearest EIA value, 0.82 *u*F, our SWL can now build his filter.

More Simple Filters. The extremely simple filter just designed for an SWL is a low-pass filter. It removes frequencies above about 3,000 Hz, and would probably make a very noticeable improvement in the noise endured by the SWL as he fished for those weak signals. If he could sacrifice some intelligibility for more freedom from noise, he could double the capacitor value, and thereby remove frequencies above 1,500 Hz.

But simplicity is not the only virtue of this filter. As a low-pass filter, it can be considered the "cousin" of three other filters—a high-pass filter, a band-pass filter, and a band-eliminate filter. All these can be derived simply and directly from this simple low-pass filter. In fact, the most common method of deriving high-pass, band-pass, and band-eliminate filters is to design first a so-called "equivalent" low-pass, and then quickly and easily derive the desired high-pass, band-pass, or band eliminate from the low-pass design. ■

Filter Design–Part 2

☐ In the first article in this series, we discussed the four basic types of filters—low-pass, high-pass, band-pass, and band-eliminate—and told what was to be expected when one of these filters was dropped into a circuit. We also showed how to measure the frequency response of a filter in a lab set-up, and we explained why the graph of frequency response is usually plotted in decibels (dB) and in decades of frequency (that is, in equal increments of 1,000 Hz, 10,000 Hz, 100,000 Hz, etc.). We then proceeded to design a simple low-pass filter, using only a resistor and a capacitor, to obtain the desired passing of low frequencies and the elimination of high frequencies.

Review of Low-Pass Filter. In our design of the simple low-pass filter we arrived at the circuit values shown in Fig. 1A and the frequency response shown in Fig. 1B.

The filter circuit in Fig. 1A was designed to have a *corner frequency* of 3,000 Hz, meaning that it would pass all frequencies *below* 3,000 Hz rather well, but eliminate those *above* this frequency. What if we desired the opposite of this effect? What if, instead, we wished to pass all frequencies *above* 3,000 Hz, and eliminate those *below* this frequency? A circuit having these characteristics is called a *high-pass filter,* and its frequency response would be as shown in Fig. 2.

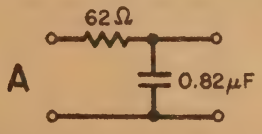
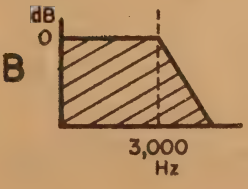


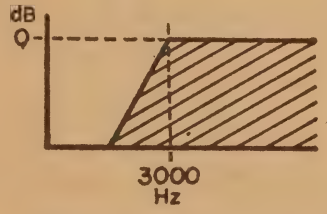
Fig. 1



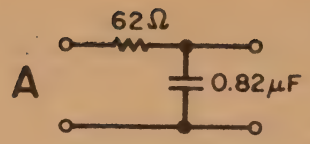
Flipped Circuit for a Flipped Response. In one sense, the frequency response of Fig. 2 is a flipped-over version of the low-pass response shown in Fig. 1A. It should not be too surprising then to learn that the circuit producing this flipped-over response is itself a flipped-over version of the low-pass filter. The two circuits are compared in Fig. 3.

If the same values are used (.82 *u*F and 62 ohms) in the flipped circuit, it yields the same corner frequency (3,000 Hz) as the low-pass design, except the roll-off (the part of the response that goes down-hill) goes down to the left instead of to the right. Just like its low-pass

Fig. 2

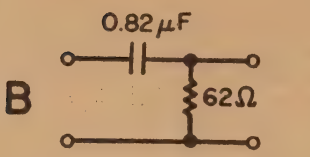


cousin, this high-pass circuit likes to be driven from a low impedance (the same 6-ohm voice coil would do nicely) and it likes to drive into a high impedance (the same 600-ohm headset would also be suitable). However, the SWL who designed the low-pass in our first article would be very disappointed in this filter. Instead of cutting out unwanted high-frequency noise to enable him to hear the signal from that distant station, this filter would cut out the signal and let him hear the noise! Of course, there are many applications where this high-



LOW PASS FILTER

Fig. 3



HIGH PASS FILTER

pass frequency response produces a desired result. If the corner were at 20 Hz, for example, it could help diminish the effects of turntable rumble in a high-fidelity system.

Designing by Flipping. If it's a high-pass filter you need, there is almost nothing to the design. You simply design the "cousin" low-pass, using the formulas in the first article, and then "flip" it to be a high-pass, as in Fig. 3B. Just choose the corner frequency for the "cousin" to be the same corner frequency that you desire in the final high-pass.

For example, let's design a high-pass filter to eliminate the hum (60 Hz) on a 600-ohm phone line before we connect it into a 100,000-ohm input of a public address (P.A.) amplifier. In this case, we would choose a corner frequency for our high-pass which was well above the 60-Hz hum, to be certain of eliminating enough of the hum. Choosing a 240-Hz corner frequency, we proceed to design the low-pass just as we did in the first article:

First, compute R as follows:

$$R = \sqrt{R_s R_L} = \sqrt{600 \times 100{,}000} = \sqrt{60{,}000{,}000}$$

$$R = 7{,}750 \text{ ohms}$$

Next, compute C:

$$C = \frac{159{,}000}{f_c R}$$

$$C = \frac{159{,}000}{240 \times 7{,}500} = 0.88\ mF$$

You would be quick to admit that the values 7.750 ohms and 0.0088 *u*F for a resistor and capacitor, respectively, are difficult to find. Therefore, select the nearest stock EIA values, which are 7,500 ohms and 0.091 *u*F.

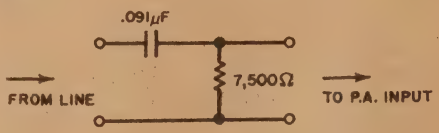
Although we designed this using the low-pass formulas, we can directly draw the high-pass circuit and plug in the calculated values. See Fig. 4.

It is here that the shortcomings of these very simple filters first become apparent. We tried to make this filter eliminate 60-Hz hum from the telephone line, while still passing voice, speech, and music well enough to yield a good-sounding audio signal. However, the roll-off—the sloping part of the frequency response—is so gradual in these very simple filters that you must start it rolling down very far above the unwanted frequency in order to make sure that it gets down far enough to be effective in eliminating the unwanted frequency. In this filter (Fig. 4) we started the roll-off on its way at 240 Hz, which is right in the middle of some very important audio frequencies. Consequently, some important low frequencies will be lost, and the output of the system will sound "tinny."

A Simple Band-Pass Filter. So far, we've discussed and designed filters which either passed low frequencies and eliminated highs, or passed the high frequencies and chopped off the low. A third type of filter, called a *band-pass* filter, passes a selected group of frequencies. Fig. 5 shows clearly how the responses of low-pass, high-pass, and band-pass filters compare.

We have established a relationship between high-pass and low-pass filters, that is, one has the *flipped-over* response of the other, and their circuits are *flipped-over* versions of each other.

Fig. 4



(Figs. 1, 2, and 3). But what is the relationship between a low-pass filter and a *band*-pass filter? We've said that all the filters are *cousins* of the low-pass filter, and can be easily derived from it. But the band-pass filter clearly has no flipped-over response, and we would expect its circuit to be quite different from its low-pass cousin.

Enter Resonance. If we make a low-pass circuit as shown in Fig. 6 we find that it passes frequencies from zero Hz (essentially DC) to its corner, 8 kHz. Since the frequencies passed occupy a band 8 kHz wide, this filter is said to have a *bandwidth* of 8 kHz. If we now put a coil across the capacitor as shown in Fig. 7, our knowledge of tuned circuits tells us that this coil and capacitor combination will *resonate*, that is, will be particularly responsive to a certain frequency, called the *resonant frequency*. It will also respond rather well to nearby frequencies slightly above and below the resonant frequency. If we graph the frequency response of this new resistor-capacitor-coil circuit, we find that we have made a band-pass filter, as can be seen from the response in Fig. 8. And here is the most important fact: since we started with a low-pass filter (Fig. 6) which had a bandwidth of 8 kHz, our new, derived band-pass *also* has a bandwidth of 8 kHz.

To derive a band-pass filter from its low-pass

Fig. 5

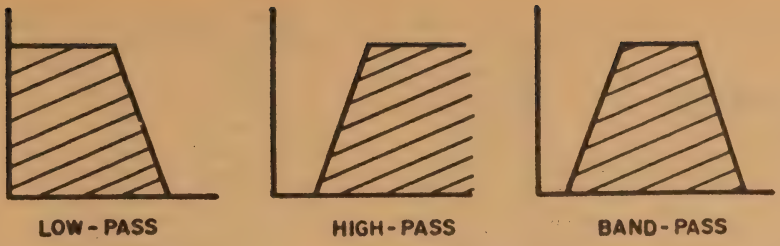
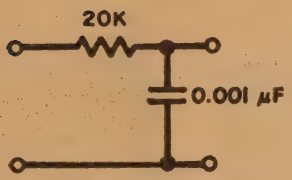
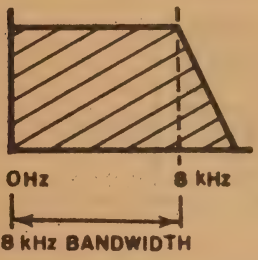


Fig. 6





cousin, then, we need only shunt the capacitor with a coil of the right inductance, and the result will be a *band-pass* filter of the *same bandwidth* as the original low-pass, but *moved up in the frequency band* to the resonant frequency of the coil/capacitor combination.

For example, let's assume we need a band-pass filter which will pass a band of frequencies from 101 kHz to 109 kHz. This means that we need a bandwidth of 8 kHz, which we determined by a subtracting 101 from 109. We therefore must first design a low-pass filter of 8 kHz bandwidth, using the formulas and methods of the first article in this series. We will assume that this has been done, and the result is the 8-kHz low-pass of Fig. 6. Now, we must convert this low-pass to a bandpass by the following steps:

1. Determine the resonant frequency needed by the formula

$$f_R = \sqrt{f_U f_L}$$

where f_U is the upper frequency of the band-pass, and
f_L is the lower frequency, and
f_R is the resonant frequency desired.

In the present example,

$$f_R = \sqrt{109 \times 101} = \sqrt{11{,}009} = 104 \text{ kHz}$$

Note that the frequencies in the formula may

Fig. 7

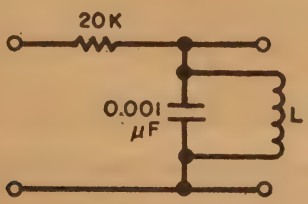


Fig. 8

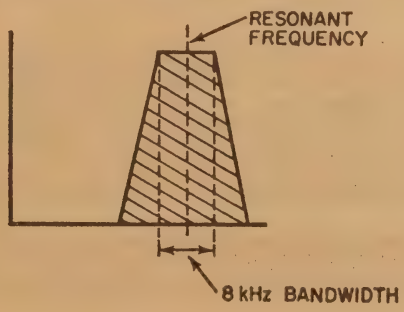
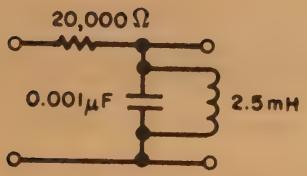


Fig. 9



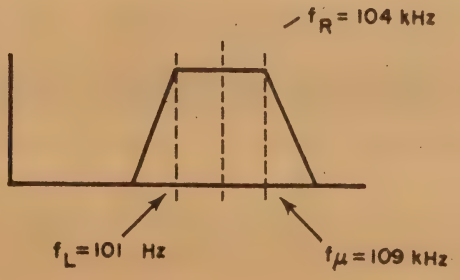


Fig. 10

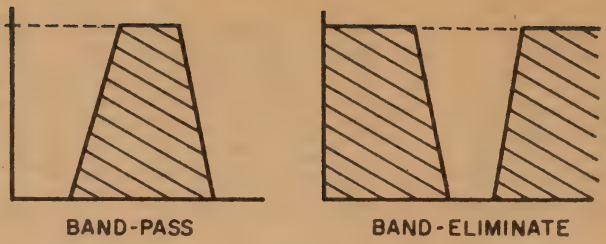
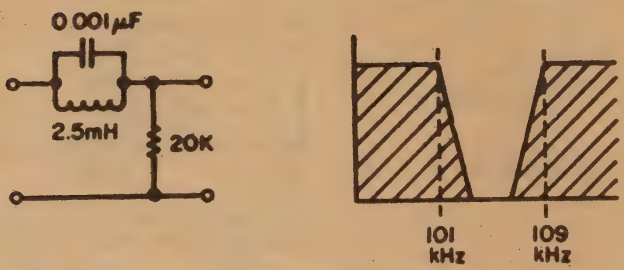


Fig. 11



be Hz, kHz, or MHz, as long as they are all the same.

2. Determine the inductance of the coil to be placed across the capacitor by the formula

$$L = \frac{1}{C}\left[\frac{159}{f_R}\right]^2$$

where L is the desired inductance in henries, C is the capacitor in the low-pass, in μF, and f_R is the resonant frequency in Hz.

In the present example,

$$L = \frac{1}{.001}\left[\frac{159}{104{,}000}\right]^2$$

$$= .0023 \text{ henry}$$

$$= 2.3 \text{ millihenry}$$

With that, the design is complete, and the circuit of the filter, with its response, is as in Fig. 9.

Only One Left. Of the four basic types of filters, we have now shown how to design three, leaving only the band-eliminate filter for the end of the discussion. If we graph the frequency response of a band-eliminate filter side-by-side with the response of the band-pass filter just designed above, we note, in Fig. 10, an interesting similarity . . . or should we say dissimilarity?

Flipped Again. The response of the band-eliminate filter is a flipped-over version of the band-pass filter, just as the high-pass was a flipped version of the low-pass. This raises in our minds the question, can the band-eliminate version of the filter be a flipped version of the band-pass circuit?

The answer is a resounding *yes!* To make the band-eliminate filter, simply interchange the locations of the resistor and the coil/capacitor combination in the bandpass, and, presto! . . . we have a band-eliminate circuit. See Fig. 11 for its circuit and response.

Our flipped version will *eliminate* the frequencies from 101 kHz to 109 kHz, and will pass all others. This is the exact opposite of the band-pass action. To design a band-eliminate filter then, you merely design a band-pass filter and interchange the components as in Fig. 11.

The filters described thus far have one thing in common: they are extremely simple, using only two to three components to perform the filtering function. Of course, it simply isn't possible to get something for nothing, so you might suspect that these very simple filters are lacking in some aspect of their performance. Such suspicions are correct; the filters are deficient in having a very slow roll-off. This means that frequencies which you would like to have totally rejected by the filter are only half-heartedly rejected. This is especially true for frequencies fairly close to the corner frequency, as pointed out in the discussion of the high-pass filter trying to reject 60-Hz hum. Even though the corner was placed at 240 Hz, the rejection of the unwanted hum was still rather poor.

In the case of band-pass circuits, the roll-offs on both sides are problems. From the shape of the response these roll-offs are sometimes referred to as "skirts"; for good performance the skirts should be as steep as possible. ■

Rapid Design Tip

Want a quick, easy method to determine what component values to use when you insert an additional transistor amplifier stage in your equipment? The common emitter arrangement shown in the diagram can be used with a reasonable amount of simplicity, stability, and dependability. It has both voltage and current gain.

Get Started. It is simple to select the correct values for the components in this circuit. Suppose we have the common variety of a silicon transistor with a beta of over 20 to be used as a small signal class A amplifier stage.

Under these conditions we want about 1 mA in the collector circuit. Let's use an emitter voltage somewhere between ½ and 2 volts, say 1 volt. The emitter resistor Re value will then be 1 volt/1 mA = 1000 ohms.

Since the base is automatically at 0.7 volt for a silicon transistor—close to 0.3 volt for germanium—the 1 volt emitter sets the base at 1.7 volts.

The current in the divider network of R1 and R2 should be ½ or more times as high as the collector current, which was selected at 1 mA. The divider current, then, is 0.5 mA. Use a 6-volt collector supply, for example. To obtain the resistance values of the divider network simply divide the 6 volt collector supply by the divider current of 0.5 mA. The answer, 12,000 ohms, is the total divider resistance value. Since R2 is at 1.7 volts/0.5 mA = 3400 ohms, it follows that R1 must be 12,000 minus 3400 or 8600 ohms.

This leaves only the collector resistor Rc to find. Any resistance should be used that gives the required collector voltage to drive the next stage. The collector voltage should be kept at least 1 volt over the emitter voltage; about 3 volts would be a logical output. The value of resistor Rc can then be easily calculated by 6 − 3 volts / 1 mA = 3000 ohms.

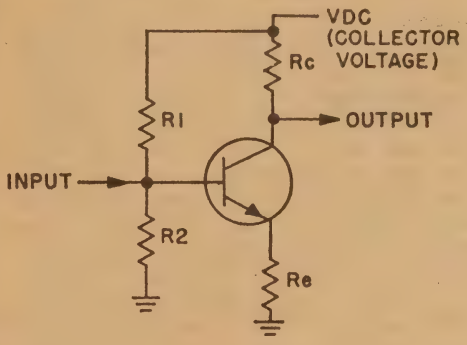
Lest You Forget. A few things worth remembering: A signal at the base affects the collector current. A positive voltage for an NPN transistor or a negative voltage for a PNP applied to the base raises the collector current. The common emitter configuration inverts the input signal 180° at the transistor output. The common emitter current gain, beta, is usually given in the manufacturer's data sheet but may quite easily be figured by the equation:

$$\text{Beta} = -\frac{\text{alpha}}{1 - \text{alpha}}$$

Thus, if alpha is given at 0.98, beta would be:

$$\text{Beta} = -\frac{1-.98}{.98} = \frac{.02}{.98} = -49$$

As a general rule alpha is approximately between .9 and .99; beta is usually between 10 and 1000. The negative number (minus sign) indicates signal phase inversion.



Final Tips. Now you have an inside track on designing simple, one-stage transistor amplifiers. If your computations call for an 8,600 ohms resistor, select the nearest standard resistance value. The two nearest values are 8,200 and 9,100 ohms. Either value is usable in the circuit (the 8,200 being the closer) since they are within 6% max. of the computed value. You'll find ±10% change in computed values to have little effect on the circuits operation. The most important part of designing an amplifier stage is to fully understand how the stage works. First important step is to actually assemble the stage on a breadboard so that you can take measurements. Check DC values across each resistor with no input AC signal. Next, apply a low level signal to the input and see what happens across resistors in the collector and emitter circuits. Use an AC-VTVM, or better, an oscilloscope. The 'scope's traces will enable you to see the input and output signal and measure overall gain visually. ■

How Microwave Ovens Work

☐ Making waves! That's the best way to describe the coming of the microwave cooking oven into America's homes. And it is a long time coming. Although not recorded in the Army's archives, a World War II tech-sergeant may have been the first person to find "consumer" microwave application when he kept his coffee warm with microwaves from an SCR-584 radar. Or was it a radio station chief engineer in the same era who kept his fried chicken warm near an FM tank circuit while Kate Smith sang *God Bless America*? And there must have been a two-letter-call ham who sizzled a weiner on a 10-meter coil following the departure of his wife to Reno.

Different kind of radiations can be grouped into two sets. First are the *ionizing radiations* such as X-rays, gamma rays, cosmic rays, ultra-violet rays which can cause chemical change to take place in foods with little or no temperature rise. Non-ionizing radiation will not chemically alter food, but will raise its temperature provided the radiation is of sufficient intensity. The non-ionizing radiations are radio wave, microwaves, infrared waves, and visible light, to name a few.

These waves also radiate outward from the center like the waves on the surface of the pond. They travel, however, at the speed of light, 186,282 miles per second, and carry small bundles of energy called photons which vibrate at various frequencies. Radiant waves are characterized by their wavelength and their frequency of vibration (number of complete cycles per second).

Wavelength × *frequency* = *the speed of light*

Thus, as the frequency increases the wavelength becomes shorter. Microwaves vibrate

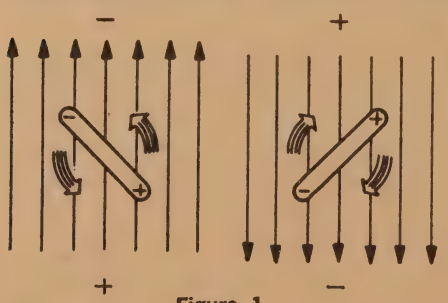


Figure 1

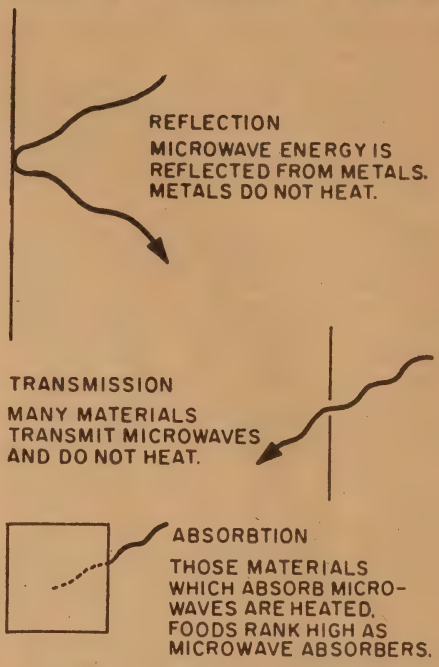

REFLECTION
MICROWAVE ENERGY IS REFLECTED FROM METALS. METALS DO NOT HEAT.
TRANSMISSION
MANY MATERIALS TRANSMIT MICROWAVES AND DO NOT HEAT.
ABSORBTION
THOSE MATERIALS WHICH ABSORB MICROWAVES ARE HEATED. FOODS RANK HIGH AS MICROWAVE ABSORBERS.

Figure 2

millions of times per second (that is, they have a very high frequency) and are, therefore, very short waves, hence the term *microwaves.*

There are two microwave frequencies in general use for microwave ovens: 915 MHz (wavelength is 32 cm or about 12.5″), and 2450 MHz (wavelength is 12 cm or about 5″). These are two of the frequencies allocated by the Federal Communications Commission (FCC) for Industrial, Scientific and Medical Use (sometimes called the ISM frequencies).

Microwave Energy Produces Heat. All matter is made up of atoms and molecules. Some of these molecules are electrically neutral, that is, they have no electrical charge. Carbon tetrachloride, benzene and parafin wax are examples of electrically neutral materials and microwave energy will pass through these compounds as if they weren't present.

Most matter *is not* electrically neutral, and when an electrical field is applied the molecules tend to behave like microscopic magnets and attempt to line up with the field. See Fig. 1. When the electric field is

reversed millions of times each second, these molecular magnets are unable to keep up because of other forces acting to slow them down. Such forces which restrict their molecular movement may be mechanical such as is the case with ice or solid fats, or viscous as is the case with a syrup like molasses. The energy of the microwaves in trying to overcome these forces is converted to heat. The material converts the energy to heat, or it might be said that the material heats itself. Another way to look at it is, consider billions of molecules rubbing elbows to keep warm.

Microwave Properties. So far two characteristics of properties of microwave energy have been mentioned: absorbtion and transmission. Like light waves, microwave energy is also reflected. See Fig. 2. Metals reflect microwaves, and since there is no absorbtion, metals *do not heat*. Many materials in addition to glass transmit microwaves, and again, since there is no absorbtion, there is no heating. Paper, china and some plastics transmit microwaves and are therefore likely candidate materials for utensils for use in microwave ovens. The overall result is that foods can be cooked or heated in an oven on utensils which are relatively cool to the touch. Hopefully, no more burnt fingers.

Fast Cooking. An additional characteristic of microwave energy is its ability to penetrate deeply into food materials and to produce heat instantaneously as it penetrates. This is in sharp contrast to conventional heating, which depends on the conduction of heat from the food surface to the inside. Conduction heating can be accelerated only by increasing the surface temperature, and obviously there are limits, for who would enjoy a roast beef that is charred on the surface and raw inside?

Measurement of the temperature distribution in an item heated in a microwave oven reveals another difference. The surface will be usually cooler than an inch or so below the surface. This is caused by radiation of

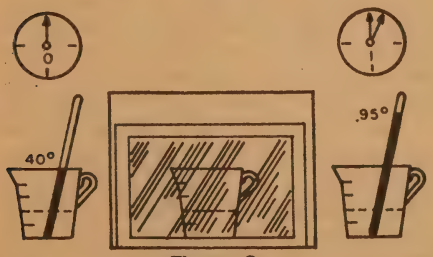


Figure 3

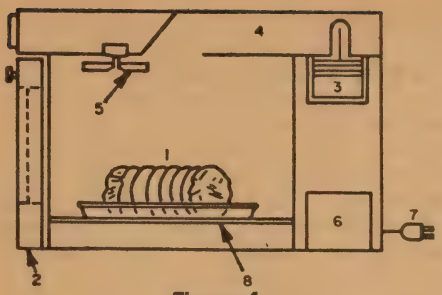


Figure 4

heat from the food surface to the cooler surroundings of the microwave oven. It does not happen in a conventional oven because the oven heat is outside the food and must slowly conduct through the food to the cooler interiors.

Power Is Heat. To properly develop a sense of timing in microwave cooking, it is first necessary to know how much power is available in the oven. A microwave oven is not like a conventional hot oven. There is no excess of heat available. All of the microwave energy is absorbed by the food. There is no wasted energy.

You can measure the power by converting microwave energy into heat and measuring it in a simple calorimeter. The tools needed are a Pyrex measuring cup, a thermometer, a clock and some cold water. See Fig. 3.

Pour two measured cups of cold tap water into a Pyrex dish or cup. Note the temperature. Heat the water in the microwave oven for exactly one minute and measure the temperature again. The difference in temperature times 17.5 is the power in watts. This is the amount of power generated in the oven in one hour. One-sixtieth (1/60th) of this power is available every minute.

A British thermal unit (Btu) is the amount of heat which will raise the temperature of one pound of water one degree Fahrenheit. In terms of Btu then, a one kilowatt oven puts out 57 Btu each minute, and if one pound of water were placed in the oven it should increase in temperature 57°F. in one minute.

Most foods contain moisture in varying amounts with the exception of pure fats like salad oils, shortening and lard. Another exception is dehydrated foods which do not depend on moisture to convert microwave energy into heat. The amount of moisture in a food has a direct bearing on its heating rate in a microwave oven.

One pound of food with 50% moisture will take less time to heat to a specific temperature than one pound of food with 75% moisture. Or if both are heated for the same length of time, the food with less water will reach a higher temperature than the food consisting largely of water. One pound of water, however, will heat only half as fast as one pound of fat. The reason for this requires an explanation of the term *specific heat.* This is a measure of a material's ability to hold heat as compared to water. Water has a value of one (1.0). Fats are about 0.5. Thus water takes twice as much heat as fat to increase its temperature by one degree. Therefore in our one kilowatt oven one pound of fat will increase in temperature by 114°F. while the same amount of water will only increase 57°F. (It should be noted that this does not apply at high temperatures where some of the water may be changed to steam. This change uses up much more energy.)

A simple formula shows the relationship between the heat required and these other factors:

"Heat required in Btu's (Q) equals weight in pounds (W) times specific heat (s) times temperature difference in degree Fahrenheit (F), or

$$Q = W \times s \times t$$

With this simple equation, you can compute how long it would take to heat 12 ounces of canned green beans to 160°F. in one kilowatt microwave oven. The specific heat of most vegetables, because of their high moisture content, is around 0.9. Let us assume that the beans would be at an initial temperature of about 70°F. (off the shelf).

$Q = 0.75$ lbs. $\times 0.9 \times (160°F. - 70°F.)$

$Q = 0.75 \times 0.9 \times 90$

$Q = 60.75$ Btu

Since our oven (1 kW) provided 57 Btu per minute, then the heat required (Q) divided by 57 Btu/minute would give the time, or

$$T = \frac{Q}{57} = 63 \text{ seconds}$$

How long would it take if the green beans were just removed from a 40°F. refrigerator?

When two foods at different starting temperatures are heated simultaneously in a microwave oven the colder food takes longer than the warmer food. An interesting example is that of apple pie and ice cream. With the pie at room temperature and a scoop of hard frozen ice cream on top, the pie can be warmed without melting the ice cream. About 15 seconds in a 1 kilowatt oven will do the trick.

Inside the Microwave Oven. Unlike most cooking appliances in which the food to be cooked is overwhelmed with heat, a microwave oven operates essentially at room temerature. In fact, if anything, the food heats the oven.

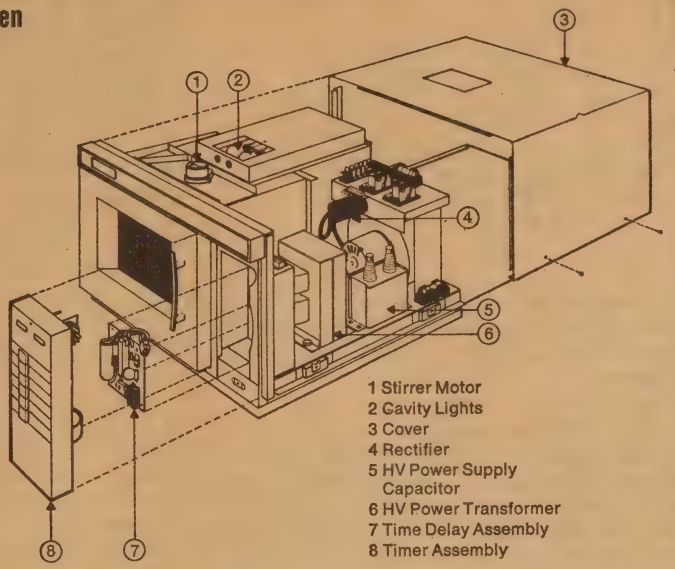
The basic parts of a microwave oven are shown in Fig. 4: the cavity (1) the door (2) the magnetron (3) waveguide (4) the mode stirrer (5) the power supply (6) and the power cord (7). The main function of the power supply is to convert low voltage line power to the high voltages required by the microwave energy generator, the magnetron. When the magnetron is energized it generates high-frequency energy which then passes down the wave-guide to the cavity. The mode stirrer as it turns interrupts the energy as it enters the cavity and causes it to be distributed more uniformly about the cavity before it is absorbed by the food load. Without the mode stirrer, standing waves would occur in the cavity and there would be regions that would receive more energy than others, causing *hot spots* and *cold spots.*

The food load is shown positioned off the floor of the oven on a glass shelf (8) for a reason. This position permits some of the energy to be reflected from the oven floor into the food from below. If this were not done the bottom of the food would lag behind in cooking giving uneven doneness.

The purpose of the oven door is to provide an access to the cavity and also to confine the microwave energy. A proper fitting door is essential to completely confine the energy and, it, as well as the door flange, should be wiped clean regularly to insure good contact of the door and the door flange. Considerable ingenuity has gone into microwave oven door design because it is such a critical feature of the oven. Good metal to metal contact is only one means whereby microwave tightness of the oven is assured. Another approach utilizes a quarter wave slot or "choke" seal around the perimeter of the door which acts to choke off or cancel out microwave emissions at this point. On ovens in which tightness is effected by a metal to metal contact, food spatters and spills on or between the

Inside a Microwave Oven

If you have seen the inside of one microwave oven you've seen them all. Basically, tracing from the line cord in, there's the rectifier, high-voltage transformer, and magnetron (hidden behind the oven's cavity). Necessary extras are stirrer assembly, oven lights and timer.



door and the flange provide a pathway for microwave energy leakage from the oven. Food buildup should not be permitted to accumulate on the door seals.

Oven controls consist mainly of a timer, indicating lights, a start or cook button and a master switch. Timers vary from one manufacturer to the next, but are usually marked so that short heating cycles can be set with some degree of accuracy. The timer is important because all microwave cooking is gauged by time, not temperature and time.

A few seconds excess time in some cases can mean the difference between success and failure. An audio signal such as a buzzer bell usually indicates when the set time has elapsed and the oven has turned itself off. The oven may also be turned off simply by opening the door. To protect the user from unnecessary exposure to microwave energy is a requirement of all microwave ovens. Two or more interlocks operate when the door is opened any one of which could turn off the oven.

The oven can also be turned off by turning the timer back to zero time or by turning off the master switch. In some oven designs the cook button is omitted and cooking action initiated simply by closing the door.

Browning. The brown surface color of foods is due to a chemical reaction between food sugars and amino acids, the building blocks of protein foods. This reaction proceeds slowly at low temperatures and is accelerated by increasing the temperature. In microwave cooking the surface temperature of foods rarely exceeds 212°F., the boiling point of water, and is usually much lower. Because of this, most foods cooked in microwave ovens lack the surface coloration expected of certain foods. A steak or a meat pattie, for example, would have a gray, unappetizing surface appearance. Baked goods would not have a brown crust. The appearance of such foods can be enhanced by placing them in a hot oven for a few minutes, or in the case of the steak and meat pattie, on a grill or under a broiler. Restaurants which do a sizable steak business presear a quantity of steaks in advance and finish them in seconds in the microwave oven when ordered. Some foods which because of their size take longer to cook in a microwave oven do take on an acceptable surface appearance. This is aided by the presence of surface fats as in beef roasts or roasting chickens and turkeys. These fats reach temperatures above 212°F. and act to accelerate the browning reaction on the meat surface.

Some Cooking Tips. Since small items cook faster than large items, it is good practice to do all of one size at a time. In baking potatoes, for example, if they are nearly alike in size they will all be finished at the same time. The alternative to size grad-

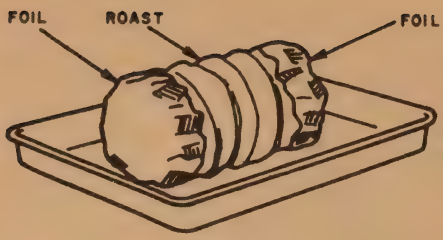


Figure 5

ing is to remove the smaller items as they are done and continue to cook the larger items. This requires considerably more attention.

Since metals, including aluminum foil, reflect microwave energy you can use this phenomenon to advantage in certain cooking operations to restrict heating in certain areas. A good example to illustrate this effect is in the cooking of a large beef roast. See Fig. 5. First, wrap aluminum foil over the outer two inches at each end of the roast and cook for 2 to 3 minutes per pound (in a 1 kW oven 20 to 30 minutes for a 10 lb. roast). The roast should be turned at least once during this time. Remove the foil and cook for an equal period of time, again turning at least once. Remove the roast from the microwave oven, cover and let stand an additional 30 to 45 minutes or until a meat thermometer inserted into the center of the roast reads 140-150°F. Shielding the roast in this way during cooking will insure a more uniform degree of doneness from one end of the roast to the other. If a more well done roast is desired, it can be returned to the oven for an additional 5 to 10 minutes.

A circle of foil placed in the center of a slice of left over roast will keep it from becoming too well done while warming it for service. When heating a casserole, the use of a strip of foil around the edge will slow down the heating effect in this area and insure a hotter center. There are many other possibilities using foil and metal forms to control microwave heating.

Regular shapes heat more uniformly in a microwave oven. When the shape is irregular the thin, narrow parts tend to overcook and may be dried out by the time the thicker parts are done. This of course, happens in conventional cooking but is less pronounced because cooking is slower. Where it is possible to control the shape, as for example with a meat loaf or by tying a beef roast into a more cylindrical form, much more uniform results are obtained. Where this is not possible, thin parts may be covered with aluminum foil for a part of the cooking cycle. The same technique can be applied in protecting the wing tips and legs of roasting chickens and turkeys.

Dinner is in the Bag. One of the principle advantages of microwave cooking is that it often can be accomplished in the serving dish or in the package in which the food was purchased. With the exception of metals, all packaging materials are transparent to microwave energy. The list includes ovenproof glass, ceramics, chinaware, plastic ware, paper containers of all types and plastic films.

It is not entirely true that in microwave cooking only the food is heated. Some plastics must be used with caution. Melamine plastic ware, for example, absorbs enough energy to cause charring in places. Such plastic ware quickly becomes too hot to handle. Styrenes give off a strong odor and deform if heated too long.

The amount of energy containers absorb can be determined by making an oven power measurement with and without the container in the oven. The difference is the amount of energy aborbed by the container.

Some ceramic dinnerware may absorb several watts of power for each ounce of weight. Considering the weight of such containers, the total power absorbed can be considerable. In some cases this is a desirable feature since foods in these dishes tend to remain warm longer after serving.

Care should be taken not to use chinaware with metallic trim designs, as the thin metal trim will tend to arc and pit, thereby damaging the appearance of the chinaware. The container is also likely to be quite hot in these regions and care should be taken in handling such dishes. It should be pointed out that no harm will come to the oven if such dishes are inadvertently used.

Save the Nutritive Value. The effect of microwave energy on vitamins in foods is negligible with a slight edge in favor of the microwave oven. Generally the effect is about the same as for conventional heating methods.

In many instances more of the natural vitamins are retained when vegetables are cooked in a microwave oven because in most cases this is done without the addition of water, thus there is no leaching out of vitamins. ■

ELECTRONICS HANDBOOK

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BASIC COURSE FOR BEGINNERS

Special Bonus Section

There is great romance, excitement, fascination and aura in electronics — and you don't have to be a graduate engineer to be involved! It makes no matter whether it be ham radio, project building, shortwave listening, CB radio, audio, or whatever interest you may have — even a new career.

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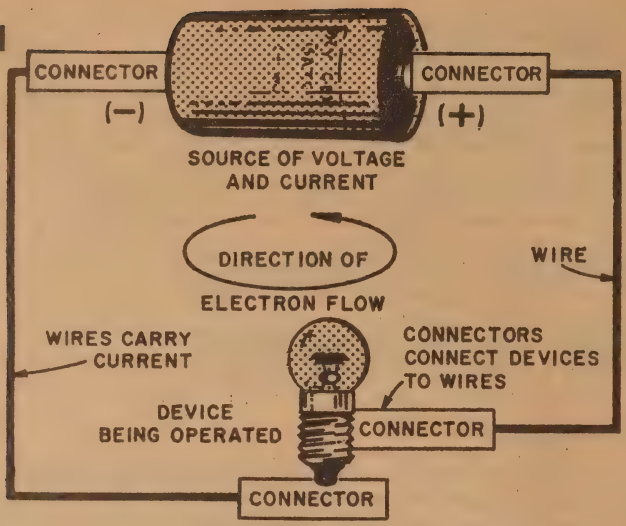
Now that you have invested $1.00 and one hour of your time, don't give up. Continue on! ELECTRONICS THEORY HANDBOOK is packed with many hours of interesting reading. And if you are well along in theory, you'll find the theory increases in scope with your interests. Do yourself a good turn. Read ELECTRONICS THEORY HANDBOOK from cover to cover.

ELECTRICAL CIRCUITS

If you look in a dictionary, you will find that *circuit* means to make a complete trip. In electricity, current makes a complete trip through an *electrical circuit.* However, if the circuit is not complete, current does not flow. Current flows only if the path through the circuit is complete. A broken wire, a loose connector, or a switch in the *off* position will prevent current from flowing.

You must now learn two important facts regarding the flow of current. A voltage source *causes* current to flow, and a complete circuit *allows* current to flow.

Voltage is measured in terms of a unit called a volt. A measurement unit indicates quantity or amount, as in gallons of water or pounds of sugar. Volts, therefore, express a quantity. Though voltage is not visible like water and sugar, the number of volts expresses an amount of electrical pressure available from an electrical power source. It is this pressure that causes current to flow from the source. The greater the pressure (number of volts), the greater the current will be through a given circuit.



A BASIC ELECTRICAL CIRCUIT

Q1. In electricity, a circuit provides a path for - - - - - - - - to make a - - - - - - - - trip.

Q2. Current flow is caused by a(an) - - - - - - - - - - - - - - and permitted by a(an) - - - - - - - - - - - - - - -.

Your Answers Should Be:

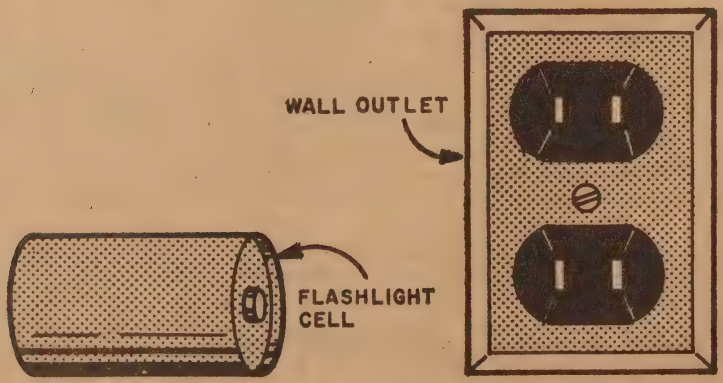
A1. In electricity, a circuit provides a path for current to make a complete trip.

A2. Current flow is caused by a voltage source and permitted by a complete circuit.

HOW ELECTRICAL CIRCUITS ARE MADE

All electrical circuits consist of the basic units shown in the drawing above. The device being operated, a small lamp in this case, can be any electrical or electronic device. In fact, many electrical circuits contain more than one device to be operated.

Now that you are familiar with the basic units of an electrical circuit, you are ready to learn more about each part.



FAMILIAR SOURCES OF VOLTAGE AND CURRENT

Voltage Source

A battery is an example of a voltage source. As you recall, a voltage source is also a source of current. See drawing bottom page 32B.

The electrical wall socket (or outlet) is another widely used source of voltage and current. The outlet is part of another circuit that has a generator as a voltage source. There may be many miles of wire between the generator and the outlet.

The battery shown in the illustration is more properly called a *cell*. A cell was originally considered to be a storage device. Cells, such as those used in a flashlight, develop approximately 1.5 volts each. A *battery*, such as the 6- to 12-volt source in an automobile, is constructed of two or more cells. In everyday English, a cell is often referred to as a battery.

Q3. List the four basic units of an electrical circuit.

Q4. A lamp will light only when it is part of a(an) - - - - - - - - - - - - - - - - - -.

Q5. A(an) - - - - - - - - - joins wires to a voltage source or operating device to make a complete current path.

Q6. A(an) - - - - - - - - - - - - - -, such as a battery, is also a source of current.

Q7. Voltage sources of the type used in a flashlight are more properly called - - - - -.

Q8. A single flashlight cell is a source of - - - volts.

Q9. If a 6-volt battery has four 1.5-volt cells, a 12-volt battery will have - - - - - 1.5-volt cells.

Your Answers Should Be:

A3. The four basic units of an electrical circuit are:
1. Voltage source.
2. Device being operated.
3. Wires.
4. Connectors (terminals).

A4. A lamp will light only when it is part of a **complete circuit.**

A5. A **connector** joins wires to a voltage source or operating device to make a complete curent path.

A6. A **voltage source**, such as a battery, is also a source of current.

A7. Voltage sources of the type used in a flashlight are more properly called **cells.**

A8. A single flashlight cell is a source of **1.5** volts.

A9. If a 6-volt battery has four 1.5-volt cells, a 12-volt battery will have **eight** 1.5-volt cells. (If a 12-volt battery provides twice as much voltage as a 6-volt battery then it must have twice as many cells.)

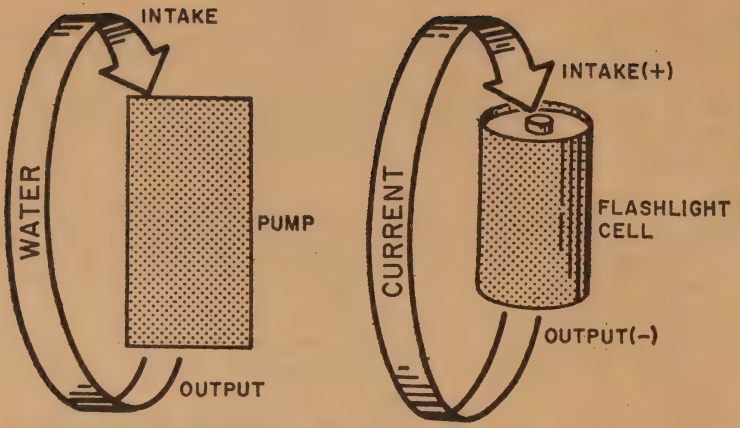
Voltage Source Connections

All sources of voltage (and current) have at least two connections.

The source of voltage and current in an electrical circuit is similar to a pump in a water system. The *pump* provides both the *pressure* and the *water* to cause a flow through the *water system*. A *voltage source* provides *electrical pressure* (voltage) and *current* (electrical equivalent of water) to cause a flow through an *electrical circuit*. See drawing on next page.

Like the water pump, *the source of voltage and current requires an input connection and an output connection.*

Safety note. Caution must always be observed when working near voltage sources or circuits. If you come in contact with both connections of the source, *your body becomes part of the circuit,* and current will flow through you. This can cause painful burns and even death. If you touch only one side of the source or a single wire leading to it, *be sure you do not touch a pipe or other metal surface in contact with the ground.* This precaution is necessary because many voltage sources have one connection wired to ground.



ELECTRICITY IS LIKE WATER IN MANY WAYS

Conductors and Insulators

Wires provide a path for electric current just as pipes provide a path for water. Metals such as copper and aluminum are most commonly used in the manufacture of electrical wire. Their atomic structures make these metals good ***conductors*** of current. Silver is the best conductor but is much more expensive than other metals. Other more economical metals, such as copper and aluminum, are good conductors and are quite easily formed into wire. When connected into a circuit, wire is most often referred to as a ***conductor.***

Most nonmetals are very poor conductors of electric current. These materials are called ***insulators.*** Rubber and plastic are two commonly used materials for insulators because they are flexible, easily molded, and can be readily cut when necessary. Because of their better insulating qualities, glass and ceramic material are used where high-voltage insulators are required.

Q10. How many connections must be made to a voltage source?

Q11. Materials which provide an easy path for current are called - - - - - - - - - - -. Those which do not provide an easy path are called - - - - - - - - - - -.

Your Answers Should Be:

A10. At least two connections must be made to a voltage source.

A11. Materials which provide an easy path for current are called conductors. Those which do not provide an easy path are called insulators.

Working With Wire

Practically all wire used in electrical and electronic work consists of a conducting metal (usually circular in cross section) covered with insulation. The insulation prevents ***undesired connections*** to and between conductors.

Exceptions include wire used for heating purposes. In these cases, the heating element (wire) is wrapped or formed on an insulating material or supported in air (a nonconductor) between insulators.

If a bare wire comes in contact with another conductor or other metal in an electrical unit, a ***short circuit*** develops. Current will flow through the ***short*** instead of the complete circuit containing the operating device. For this reason, wire should be handled with sufficient care to ensure that its insulation is not damaged.

Wire Stripping. In order to join a wire to a connector, a length of insulation must be removed from the wire. A metal-to-metal connection is required to

permit current flow.

The process of removing the insulation is called *wire stripping*. Both wire cutting and stripping are usually done with a type of pliers called *diagonal cutters*.

Precautions. When stripping wire, do not be discouraged if at first you cut the end of the wire while stripping it. Success will come with practice.

Do not squeeze the plier handles too tightly when attempting to remove the insulation from the wire. Just break the surface of the insulation with the cutting head. The cut need not go through to the wire. A steady pull should then part or tear the remaining insulation. Placing the index finger between the handles prevents the cutters from closing completely and nicking or cutting the wire.

Q12. Nearly all metals will conduct - - - - - - -.

Q13. Copper or aluminum are used in electric wires because they are good - - - - - - - - - -.

Q14. Materials that are nonconductors of current are called - - - - - - - - - -.

Q15. Insulation is used on wires to (make, prevent) contact with other conductors of current.

Q16. Undesired contact between two conductors is caller a(an) - - - - - - - - - -.

Q17. - - - - - - - - - - is the process of removing insulation from a wire. It can be accomplished by pliers known as - - - - - - - - - - - - - - -.

Your Answers Should Be:

A12. Nearly all metals will conduct current.

A13. Copper or aluminum are used in electric wires because they are good conductors.

A14. Materials that are nonconductors of current are called insulators.

A15. Insulation is used on wires to prevent contact with other conductors of current.

A16. Undesired contact between two conductors is called a short circuit.

A17. Wire stripping is the process of removing insulation from a wire. It can be accomplished by pliers known as diagonal cutters.

Devices

Current from a voltage source operates devices such as electric light bulbs, heaters, and motors. Radio and television receivers are also operated by current from voltage sources. These devices process voltages and current contained in received radio waves by changing the input energy into sound and pictures. The voltage sources make it possible for these devices to perform this process.

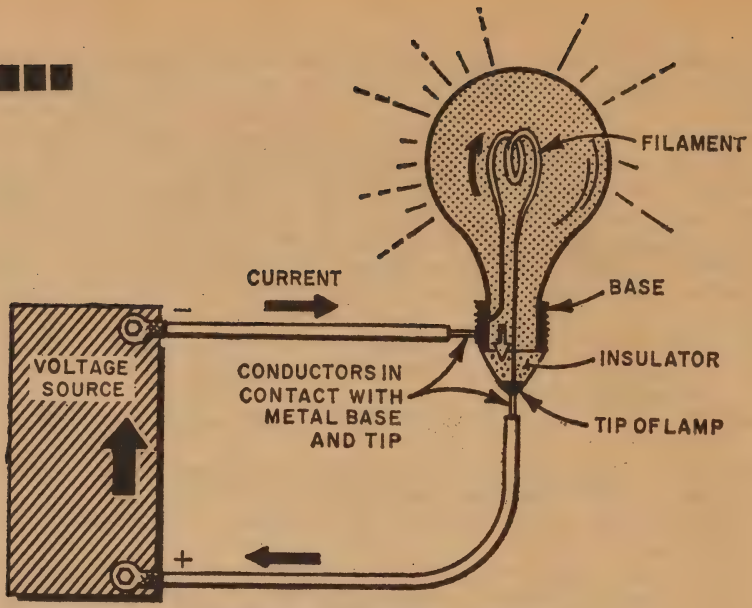
Connections. As stated earlier, all electrical devices must have *two or more* connections to a circuit. These connections are used to join conductors to the device, thus completing the circuit and permitting current to flow into and out of the device.

Operation. The voltage source operates the device by forcing current through the circuit. All connections must be made in the circuit, including those at the device and the source. Current will then be able to flow through the device and cause it to operate.

Connectors

The terms *connectors* and *terminals* are often used interchangeably. A *connector*, however, is normally thought of as being a mechanical part, such as a battery clamp, used to connect a conductor to a device. A *terminal*, on the other hand, is a point on a device where a connection can be made—a screw or other contact point.

The drawing (next page) shows how connections are made to a voltage source and an operating device. The lamp will light with the bare conductors merely touching the lamp terminals. In practice, however, the lamp is placed in a



A PRACTICAL ELECTRICAL CIRCUIT

socket and the wires connected to the socket terminals.

Wires, connectors, and terminals allow current to flow in a circuit because they are made of conducting metals. Care must be taken, however, when joining these parts to each other. Metal at the contact points must be clean and free from the insulating properties of dirt, grease, etc. Sandpaper or a small file can be used to clean these junction points when necessary. After a wire has been stripped, it should be cleaned of any remaining insulation.

When connecting a wire to a terminal, make sure the screw or clamp makes a tight connection. For current to flow, all parts of the circuit must be connected.

Q18. To permit current to flow into and out of a device, the device must have at least - - - connections.

Q19. That part of a device where a connection can be made is called a - - - - - - - -.

Your Answers Should Be:

A18. To permit current to flow into and out of a device the device must have at least two connections.

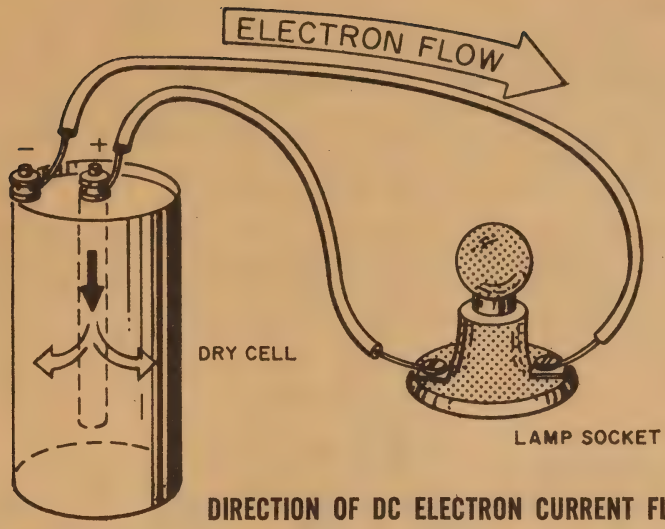
A19. That part of a device where a connection can be made is called a terminal.

A PRACTICAL CIRCUIT

The circuit shown next demonstrates the way in which all basic circuits are connected. It contains a voltage source, wires, connectors (or terminals), and an operating device. The voltage source pictured is a large 1.5-volt dry cell used in some doorbell systems. This is a practical circuit because it will actually work and is often used.

Open Circuits

If all the connections are made as shown in the illustration, the lamp will light. If any of the connections are not properly made, the lamp will not light —a condition known as an *open circuit.* An open circuit represents a condition that prevents the flow of current. In other words, the circuit is not complete.



DIRECTION OF DC ELECTRON CURRENT FLOW

Closed Circuits

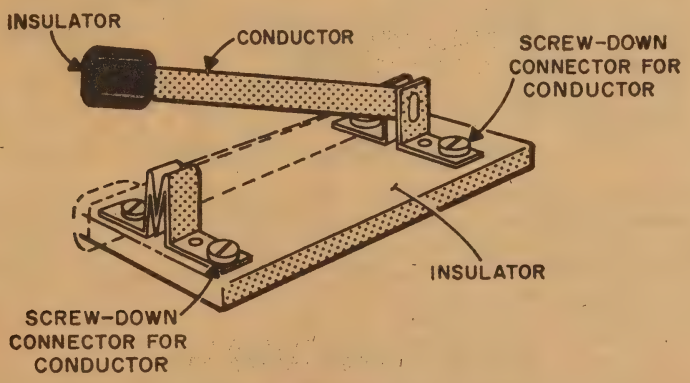
A *closed circuit* has all of its connections made and forms a complete path through which current can flow.

SWITCHES

Since it is often desirable to open and close a circuit, nearly all circuits contain some form of *switch.*

Knife Switch

The simplest type of switch is called a *knife switch.* It was given this name because it has an element resembling the blade of a knife.



AN ELECTRICAL KNIFE SWITCH

Q20. The lamp in the illustration on the opposite page lights because is it a(an) (open, closed) circuit.

Q21. Disconnecting one of the wires will develop a(an) (open, closed) circuit.

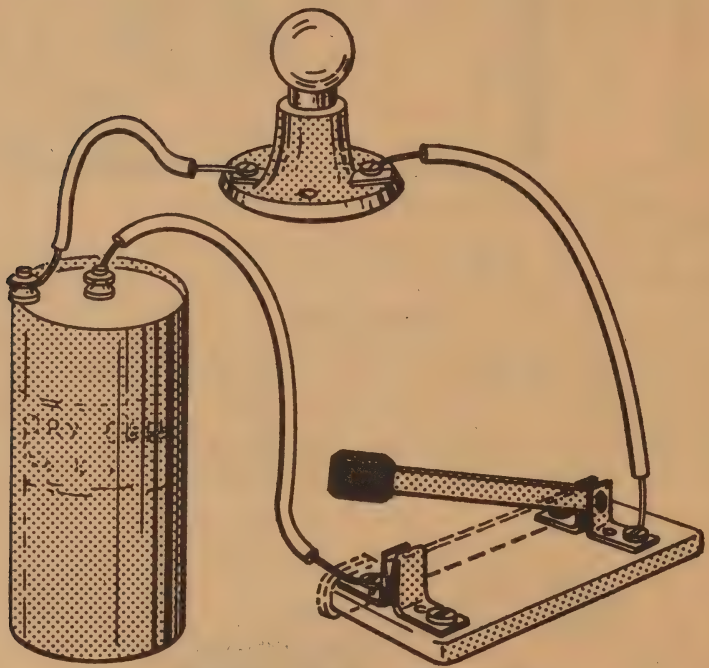
Q22. To permit opening and closing a circuit, a(an) - - - - - - can be connected into it.

Q23. A(an) - - - - - - - - - - is the basic type of switch.

Your Answers Should Be:

A20. The lamp lights because it is in a closed circuit.
A21. Disconnecting one of the wires will develop an open circuit.
A22. To permit opening and closing a circuit, a switch can be connected into it.
A23. A knife switch is the basic type of switch.

The basic circuit just explained can be reconnected to include a knife switch. The illustration shows how the connections are made. Be sure you understand what happens to the flow of current when the switch is open (position shown) and when it is closed.



A PRACTICAL CIRCUIT WITH SWITCH

There are many other types of switches, some of which you have used. For example, there are switches on the walls of your home, on the front of your appliances, and on the dashboard of your car. Nearly all operate on the knife-switch principle. In the closed position, a metal blade makes an electrical contact between at least two conductors.

VOLTAGE AND CURRENT

In this section you will become acquainted with voltage and current measurement units. You will also become familiar with the commonly used values of these units.

Voltage

Voltage is measured in terms of a unit called a volt; the number of volts expresses the amount of electrical pressure available from the source. As you remember, it is this pressure that causes current to flow. The greater the pressure (number of volts), the greater the current will be.

Current

Current is measured in terms of a unit called an *ampere*. The number of amperes defines the amount of current that is flowing in a circuit. A typical flashlight lamp (bulb), for example, draws 0.25 *ampere* (abbreviated as A or amp) from the voltage source.

A 100-watt lamp draws approximately 1 amp from the 117-volt home electrical system. Ten amps flow through some electric irons, toasters, and heaters. A car battery supplies 100 amps or more to a starter motor.

Large and Small Values

Values of voltage and current can be very large or very small. Since it is awkward to talk and write about 500,000 volts or 0.003 amp., units which are more easily handled have been developed. With this system the quantities mentioned become 500 kilovolts and 3 milliamps, respectively. A *kilovolt* represents 1000 volts and a *milliamp*, 0.001 amp.

The table will help you convert from one unit to another.

CONVERSION TABLE

When You See	Do This to Convert	Example
Mega or M	Multiply by 1,000,000	2 Megavolts is 2,000,000 volts
Kilo or k	Multiply by 1,000	5 Kiloamps is 5,000 amps
Milli or m	Divide by 1,000	7 Millivolts is 0.007 volt
Micro or μ	Divide by 1,000,000	9 μampsis 0.000009 amp
Nano or n	Divide by 1,000,000,000	5 nano volts is 0.000000005 volt
Pico or p	Divide by 1,000,000,000,000	4 pico-amps is 0.000000000004 amp

Q24. Voltage is measured by a unit called a(an) - - - -

Q25. The number of volts indicates the quantity of - - - - - - - - - - - - - - - - - - contained in a voltage source.

Q26. An ampere is a unit that indicates the quantity of - - - - - - - .

Q27. Assuming that the voltage source can provide the current, what determines the number of amps that will flow in a circuit?

Q28. 3 kilovolts is (larger, smaller) than 100 millivolts.

Q29. How much of an amp is 15 microamps?

Q30. Convert 16 megavolts to volts.

Your Answers Should Be:

A24. Voltage is measured by a unit called a volt.

A25. The number of volts indicates the quantity of electrical pressure contained in a voltage source.

A26. An ampere is a unit that indicates the quantity of current.

A27. The operating device determines the number of amps that will flow in a circuit, assuming the voltage source can provide it.

A28. 3 kilovolts is larger than 100 millivolts.

A29. 15 microamps is 0.000015 amp.

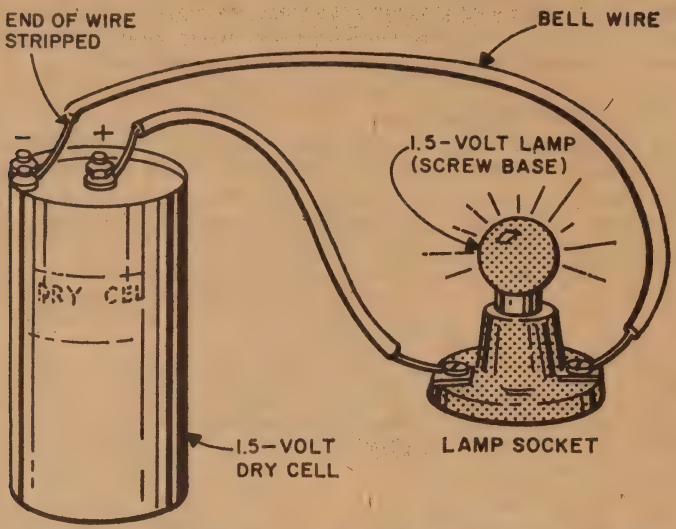
A30. 16 megavolts is equivalent to 16,000,000 volts.

DIRECT CURRENT

A current that always flows in the same direction is called *direct current*. Dry cells and batteries are sources of direct current. Some type of electric generators also supply direct current. Eventually you will learn about power supplies which provides direct current for use within radio and TV receivers.

Is There a Direct Voltage?

Yes. A voltage which provides direct current is considered to be a direct voltage. Since direct current is abbreviated DC, the abbreviation is used to identify direct voltage as DC voltage. Direct current is often shortened to DC current, or merely DC.



CONNECTION AND OPERATION OF AN ELECTRICAL DEVICE

Direction of Current Flow

Marking the terminals of a voltage source with plus (+) and minus (−) signs indicates the direction in which current flows in a circuit. There are two systems describing the direction of current flow—*conventional* and *electron*.

The conventional current theory was the first to be developed. Benjamin Franklin is considered to be its originator, and it is still being used in many electrical engineering texts. ***Conventional current*** is said to flow ***from the positive*** (+) voltage terminal, through the circuit, and to return ***to the negative*** (−) voltage terminal.

The ***electron current*** theory, of more recent origin, permits a clearer explanation of how current flows through electronic circuits. For this reason, the electron current direction of flow will be used in this course. This theory states that current ***leaves the negative*** (−) terminal, flows through the circuit, and ***returns to the positive*** (+) terminal of the voltage source.

Current flow does all the work involved in the operation of any electrical or electronic device, whether it is a simple lamp or a complicated electronic system on the Moon. In any application a continuous path must be provided between the two terminals of a voltage source before current can flow.

Q31. The connecting posts on the cell in the illustration are marked (+) and (−). The (+) post is the - - - - - - - - terminal.

Q32. Inside the cell, electron current flows from the - - - - - - - - terminal to the - - - - - - - -.

Your Answers Should Be:

A31. The (+) post is the positive terminal.

A32. Inside the cell, electron current flows from the positive terminal to the negative terminal. Though the text did not provide this information, the illustration reveals the proper direction.

Current must flow in this direction inside the battery if it is to move from negative to positive through the circuit.

ALTERNATING CURRENT

A current that reverses its direction of flow at regular intervals is called *alternating current (AC)*. You might ask, "Why should we have a current that is constantly changing its direction?" The answer is fairly simple. AC has certain features that make it desirable. The two main reasons are:

Reason 1. Wall outlets in your home supply an AC voltage. This voltage is produced by generators located many miles away. During the earliest days of electricity, DC was supplied to homes. However, DC can be sent through lines for only short distances.

AC can be easily changed to a higher or lower value. This characteristic makes possible its economical transmission over long distances—hundreds of miles in some cases. As a result, AC generating plants can be located at remote sources of water power and still be able to supply customers miles away. A good example of this application is the generating equipment at Hoover Dam in Arizona supplying power to cities on the West Coast, hundreds of miles distant.

Reason 2. Energy in many forms, such as sound and radio waves, occurs in alternating cycles. Sound waves, for instance, are alternating areas of maximum and minimum air pressure. When converted into electricity, as in the telephone, the resulting current is also alternating, thus the sound is faithfully transmitted. Clearly, it simply isn't feasible to convey such information via DC.

WHAT LIMITS CURRENT FLOW?

You have learned that voltage is a pressure which forces current to flow through a circuit. You also have learned that current has the ability to heat a lamp filament white-hot and thus produce light. But have you ever wondered why a 40-watt lamp produces less light than one rated at 100 watts? After all, the amount of voltage pushing current through both lamps is the same. The answer, of course, is the individual characteristic of each lamp which limits the amount of current that will flow.

The 100-watt lamp glows more brightly because more current is allowed to pass through the filament, heating it to a higher degree, thus causing it to give off more light. Less current is allowed to flow through the 40-watt filament. The reason for the different amount of current through each of the two lamps is an electrical characteristic called ***resistance.*** Resistance ***limits*** or ***controls*** the flow of current.

WHAT IS RESISTANCE?

Resistance is a physical property of all materials and is directly responsible for the amount of current which will flow through a material with a given voltage applied.

Atomic Structure

All matter is made up of invisible particles called *atoms.* There are over 100 different atoms, or *elements,* as the physicist calls them. One of the features that makes one atom different from another is the number of *electrons* each contains. A hydrogen atom has one electron, an oxygen atom has eight, and a uranium atom has 92.

You know that current is a flow of electrons and that electrons are made to move by a voltage. This does not mean that an electron leaves the negative pole of a battery and speeds around the circuit to the positive terminal. Instead, there is a general movement or drift of electrons throughout the complete circuit.

The next illustration shows a greatly magnified and exaggerated view of a length of wire with four atoms—A, B, C, and D. Actually, in the shortest possible length of a very thin wire there are many millions of atoms.

ELECTRON MOVEMENT IN A CONDUCTOR

Electron Flow

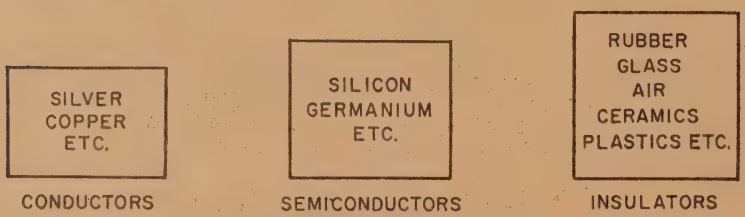
As shown in the illustration, electrons orbit about the center of an atom. At the instant voltage is applied, two things happen simultaneously—negative voltage at one end of the wire pushes against the electrons, and positive voltage at the other end of the wire pulls them toward that end. In moving, electrons strike other electrons. One electron is bumped out of atom, A, and it in turn pushes another out of atom B. At the positive end, an electron is pulled from atom D and another leaves atom C to replace it. The atoms of some materials give up their electrons more easily than the atoms of other materials.

Resistance of Materials

There is no perfect conductor. Even the best conductors, such as those having silver or copper atoms, resist the pressure to release electrons. On the other hand, the best insulators have atoms which, under conditions of sufficiently high voltage, give up some electrons. The resistance of a material, then, is ***determined by its atomic structure.***

The size of the columns in the illustration shows the comparative resistance of certain materials. Keep in mind that no material is a perfect conductor or a perfect insulator.

Most metals contain atoms that release electrons very easily. These materials, therefore, offer the least resistance to current flow. Insulators have the greatest resistance because their atoms resist the release of electrons. The in-between materials are neither good conductors nor insulators. Among these semi-conductors are the materials from which ***transistors*** are manufactured.



RESISTANCE OF MATERIALS

Unit of Measurement

Resistance is measured in *ohms;* the device for conducting such measurement is called an *ohmmeter*. The resistance of a 1.5-volt lamp, for example, is approximately 6 ohms. In other words, the lamp offers 6 ohms of resistance to the electrical pressure of a 1.5-volt cell, and the result is a current flow of 0.25 amp.

Q33. What is the difference between a conductor and an insulator?

Q34. The resistance of a material is determined by its - - - - - - structure.

Your Answers Should Be:

A33. Conductor atoms give up their electrons more easily than insulator atoms.

A34. The resistance of a material is determined by its atomic structure.

Volts, Ohms, and Amperes

Since resistance limits the amount of current that flows and voltage forces an amount to flow, there must be some numerical relationship between them.

You would see current decrease to half its former amount when a second lamp is added in series with the lamp circuit just mentioned. Current is divided by two when resistance is multiplied by two. Mathematicians say, then, that current is inversely proportional to resistance. In other words, ***current decreases*** by the same amount that ***resistance increases.***

You can also discover, by experimenting, what happens to current when voltage increases. You will find they increase together (they are directly proportional to each other). This makes sense because the pressure of voltage causes current to flow. If the pressure increases, flow increases.

These relationships of voltage and resistance to current can be expressed in an arithmetic statement as:

$$\text{Current in a circuit} = \frac{\text{voltage applied to a circuit}}{\text{resistance of a circuit}}$$

Using mathematical symbols, this statement becomes:

$$I = \frac{E}{R}$$

where,

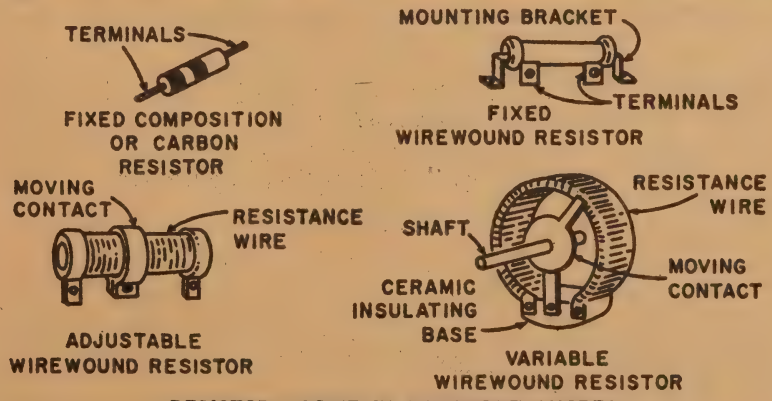
I is the current in amperes,
E is the voltage in volts,
R is the resistance in ohms.

If voltage *(E)* is increased, current *(I)* will increase. When E decreases, I also decreases. The relationship between I and R is just the reverse. A decrease in R causes I to increase. The larger R becomes, the smaller I will be. The formula above is known as *Ohm's law.*

TYPES OF RESISTORS

Resistors are classified in two ways: 1) in terms of their construction (wirewound and composition); 2) in terms of their type or function (fixed, adjustable, variable).

Wirewound resistors are made by wrapping resistance wire around a ceramic



RESISTORS COME IN DIFFERENT SHAPES

or other high-insulation cylinder. The assembly is then covered with enamel glaze and baked. The wire has a known value of ohms per inch. The resistance value desired is then merely a matter of wrapping on the required length of wire.

Composition or Carbon Resistors

Composition or carbon resistors are molded from a paste consisting of carbon (a conducting material) and a filler. Terminal wires (sometimes called *pigtails*) are inserted into the paste before it hardens. The resistor is then covered with a plastic coating. The resistance of a composition resistor is determined by the ingredients (percentage of carbon) and its diameter and length.

Fixed Resistors

A fixed resistor has only one nonvariable ohmic value.

Adjustable Resistors

Adjustable resistors provide a range of resistance within the limits of their total value. When placed in a circuit, the sliding contact can be positioned and secured to accurately provide the required resistance value. This type of resistor is not designed to be continuously variable.

Variable Resistors

Variable resistors are designed for continuous adjustment. A shaft to control the resistance value is usually connected to a knob on the front panel of an electrical or electronic device. The volume control of your radio or TV set is an example.

A rheostat is a variable resistor. The material that the moving contact presses against may be either resistance wire or a carbon mixture.

Typical applications for each kind of resistor are presented below.

RESISTOR APPLICATIONS

Type	Applications
Composition or Carbon	Composition resistors are the least expensive of the types discussed. They are, therefore, the type most widely used. However, composition resistors have certain limitations. They cannot handle large currents, and their measured values may vary as much as 20% from their rated resistance.
Wirewound	Wirewound resistors are more expensive to manufacture. They are used in circuits which carry large currents or in circuits where accurate resistance values are required. Wirewound resistors can be made to within 99% or better of the desired value.

Q35. Composition resistors are made from a - - - - - - and - - - - - - mixture.

Q36. The control that dims the dashboard lights in an automobile is a(an) - - - - - - - - resistor.

Q37. - - - - - and - - - - - - - - - - resistors are not designed to be continuously variable.

Your Answers Should Be:

A35. Composition resistors are made from a carbon and filler mixture.

A36. The control that dims the dashboard lights in an automobile is a variable resistor.

A37. Fixed and adjustable resistors are not designed to be continuously variable.

RESISTOR POWER RATINGS

As you already know, current passing through a resistor generates heat. If too much heat is generated, the resistor will be damaged. Wire in the wound resistor will melt and become open, or some of the carbon in the composition resistor will burn away.

The current-carrying capacity of a resistor is rated according to the amount of heat it can safely release in a given period of time. A resistor cannot be used in a circuit where current causes heat to build up faster than the resistor can dissipate it. When such a condition exists, the resistor may become so hot that it will be destroyed. Even if the resistor doesn't melt and become open, the excessive heat may cause a permanent change in its resistance value. In addition, heat from the overloaded resistor may damage other components that are near by.

Since heat is a form of energy, the heat-releasing rate of a resistor is measured in energy units. The unit is a *watt*. A 100-watt lamp dissipates 100 watts of heat. In the process, the lamp also gives off light.

Heat energy depends on the amount of current flowing through a resistor. The arithmetic involved is:

$$\text{Heat energy in watts} = (\text{current in amps})^2 \times (\text{resistance})$$

This means that the number of watts dissipated by a resistor can be found by multiplying the resistance in ohms times the square (a number multiplied by itself) of the current in amperes. The electrical term for heat energy is *power*.

For example, a 10-ohm resistor has three amps flowing through it. What must be its power rating in watts?

$$\text{Power} = (\text{amps})^2 \times (\text{ohms}) = (3)^2 \times (10) = 90 \text{ watts}$$

Composition resistors usually come in power ratings of ¼ watt, ½ watt, 1 watt, and 2 watts. If larger power ratings are required, wirewound resistors are used.

A design engineer determines the value of resistance needed and the amount of current that will flow through it. He then specifies the resistor wattage that must be used. If the value falls between two of the ratings mentioned above, he selects the higher rating.

Q38. Which of the standard composition-resistor ratings would you select for a resistor of 10 ohms through which 1/10 of an amp flows?

Q39. A 1-watt wirewound resistor (will, will not) safely carry more current that a 2-watt composition resistor of the same resistence.

Your Answers Should Be:

A38. $\text{Power} = \left(\frac{1}{10}\right)^2 \times 10 = \frac{1}{100} \times 10 = \frac{1}{10}$ watt

The next highest standard rating is a ¼-*watt* resistor.

A39. A 1-watt wirewound resistor will not safely carry more current than a 2-watt composition resistor of the same resistance.

RESISTOR TOLERANCE

As mentioned previously, a resistor will rarely measure the exact number of ohms specified by its label. The amount it will vary is called *tolerance*. Every resistor has a tolerance rating.

Resistor tolerance is given as a ***percentage value*** which indicates the amount that a resistor may vary above or below its labeled value. Standard tolerances for composition resistors are 5%, 10%, and 20%. Wirewound resistors may have tolerances as low as 1 or 2 percent.

Try a 1000-ohm, 10% tolerance resistor as an example. Ten percent of 1000 is 100 ohms. The tolerance factor thus indicates this resistor will measure somewhere between 100 ohms above and 100 ohms below the labeled value of 1000 ohms. This is a range from 900 to 1100 ohms. The same resistor with a 20% tolerance will have a true ohmic value somewhere between 800 and 1200 ohms.

WHAT YOU HAVE LEARNED

1. An electrical circuit provides a complete path for current flow.
2. Every electrical circuit consists of: (1) a source of voltage which causes current to flow; (2) conductors which provide a path for the current; (3) electrical devices which are operated by the current; (4) connectors (terminals) to join conductors to a source or a device.
3. Volt and ampere quantities are often expressed in very large and very small numbers. To ease the task of writing or speaking of very large or very small numbers, prefixes, such as mega-, kilo-, milli-, and micro-, have been added to the basic units of volts and amperes.
4. A current that always flows in the same direction is direct current. Its abbreviation is DC, which can be used to specify DC current or DC voltage.
5. Current flows from the negative terminal of a voltage source, through the circuit, and returns to the positive terminal. Inside the voltage source, current flows from the positive to the negative terminal.
6. A current that reverses its direction of flow at regular intervals is called alternating current (AC). AC voltage and current can be transmitted over long distances, but DC cannot. AC is also the only means of converting certain types of energy into useful electrical representations.
7. Resistance is a property of all materials which limits the flow of current.
8. Conductors have a low resistance; insulators have a high resistance.
9. Since voltage causes a certain amount of current to flow and resistance limits the amount that will flow, there is a special relationship between current, voltage, and resistance. This relationship is expressed by the following:

$$I = \frac{E}{R}, \text{ or current} = \frac{\text{voltage}}{\text{resistance}}$$

$$R = \frac{E}{I}, \text{ or resistance} = \frac{\text{voltage}}{\text{current}}$$

$$E = IR, \text{ or voltage} = \text{current} \times \text{resistance}$$

10. The unit of resistance is the ohm. The value of resistance in ohms can be measured with an ohmmeter.
11. Current flowing through a resistance generates heat.
12. Resistors are designated by construction (wirewound or composition) and by intended use (fixed, adjustable, or variable).
13. Resistances are rated by their heat-dissipating capability in terms of watts.
14. Resistor tolerance is given as a percentage value which indicates the amount a resistor may vary above or below the labeled value.

This section is based on material appearing in Vol. 1 of the 5-volume set, BASIC ELECTRICITY/ELECTRONICS, published by Howard W. Sams & Co., Inc. @ $22.50. For information on the complete set, write the publisher at 4300 West 62nd St., Indianapolis, Ind. 46268.

How Superhets Grew Up

□ Major Edwin Armstrong was too busy to take time to contemplate the superior performance of his newly developed radio receiver which was spread out on the laboratory table before him. He and his assistants had been in France only a short time but they had already witnessed the war devastation wrought upon the countryside. But at this time they were probably too far from the front lines to feel the earth-shaking rumble of the German Army Big Bertha guns which gave evidence of the last desperate attempt by the enemy to stem the sweep of the American Expeditionary Forces across France. The year was 1918 and World War I would soon be drawing to a close.

Armstrong called his new radio a superheterodyne and his genius helped save the day for the Allies. When American troops jubilantly landed on French soil they were hopelessly unprepared with any type of electronics which would intercept those faint German communications signals. Armstrong, with a host of significant radio developments already to his credit, was commissioned to go to France and solve the problem.

Those first scenes in that French radio laboratory with its vast array of radio components and with those large electron tubes glowing like embers under a witches' caldron, must have looked as impressive as a Saturn V launch. But the code signals that filled the room were a far cry from what was to come—for the days of radio broadcasting as we know them were a mere two years away. Little did Armstrong realize that he had created the most important radio circuit the world had ever seen. For every present day TV set or radio bears his mark—they all use that same basic circuit concept born in the mind of this great radio genius.

At the Beginning. Let's take a look at some of the things that led Armstrong down the path of the superhet. First of all, the early radio designers quickly found out that it wasn't easy to build an RF amplifier that would work at high frequency. By high frequency I mean 1500 kHz —the top of the broadcast band! This was mostly blamed on the tubes available at that time, triodes, which had so much interelectrode capacity that the plate energy would sneak back across to the grid and force the circuit break into oscillation. At the low end of the broadcast band, the RF amplifier was OK; but as the frequency increased, so did the whistles and screeches.

Armstrong reasoned that if the tubes worked well at low frequency, why not somehow convert the high frequency antenna currents to a low frequency within the radio receiver. He already knew that if two RF signals were mixed together their frequencies magically add and subtract to produce, among other things, a frequency which is the difference frequency between the two. Why not make some kind of an electronic *lash-up* which would mix the incoming antenna signal with one generated in the radio set to produce a low frequency difference? Then follow this up with a narrow band RF amplifier that would amplify this nice low frequency without problems with oscillation.

Well, the Major did just that and let's take a look at one of Armstrong's first circuits. This is shown in simplified form in Fig. 1 and we see the oscillator tube V8 with its associated components. It is made to oscillate by feeding plate

A completely restored superheterodyne receiver, a 1924 Radiola, was the result of the pioneering efforts and developments of Major Edwin Armstrong and his early staff.

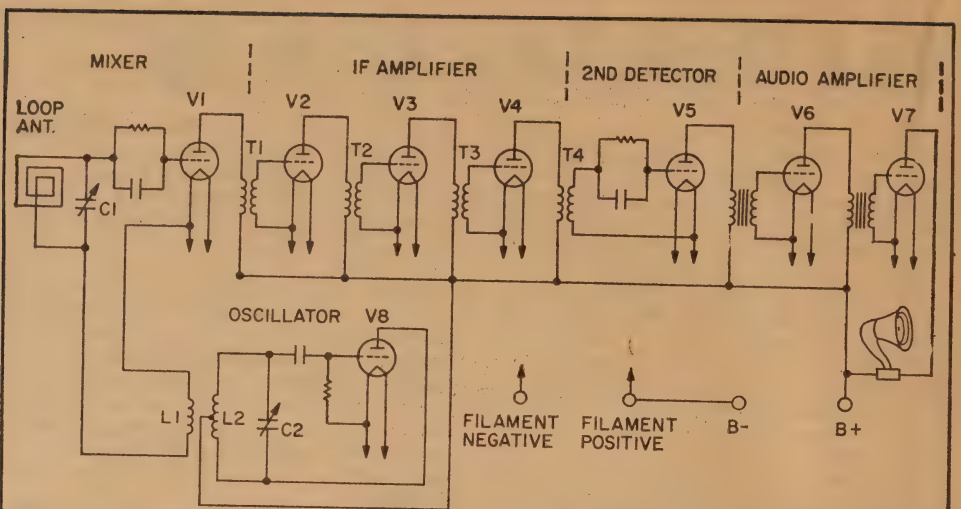


Fig. 1. A very early superheterodyne circuit varies very little in basis from today's circuits used in modern receivers. The local oscillator stage, V8, pumps the mixing signal to the antenna. A great idea at the time, but the circuit radiated some of the signal into the ether. This would cause considerable noise and jamming with today's billions of receivers.

energy back to grid circuit and tuned by the L2-C2 combination. These oscillation currents are induced in the pickup coil L1 which is connected in series with the loop antenna. This arrangement allows the incoming signal from the loop, tuned by C1, to be mixed with the oscillator frequency tuned by C2 and fed to the control grid of V1—the mixer stage. The voltage variations in the plate circuit of V1 are a mixture of the radio station's frequency plus those produced by the oscillator which combine to produce the difference of the two.

How It's Done. The mixing operation can go something like this. If C1 (See Fig. 1) is tuned to the station frequency of 1000 kHz and C2 to 1100 kHz, the difference frequency from the mixer is 100 kHz. Transformers T1, T2, and T3 are wound to resonate at 100 kHz and, together with V2, V3, and V4, constitute an amplifier adapted to amplify this 100 kHz difference signal. This section of the receiver was called the intermediate frequency amplifier, or IF amp. After demodulation in V5 and after going through audio amplifiers V6 and V7, out comes the voice program. For every new setting of the mixer tuning, the oscillator is tuned exactly 100 kHz above it. So the purpose was accomplished—the mixer-oscillator combination converted every incoming signal to a nice comfortable low value that made the IF amplifier feel at home.

Results. The circuit's sensitivity and selectivity proved to be astounding. Coast to coast reception became a reality with the *superhet.* One proud owner of that period bragged that his set would amplify a weak signal so well that if a fly walked on the antenna wire it would come out of the speaker sounding like a jungle war dance.

Problems. With the coming of the radio broadcasting era in 1920, you would think that the superhet would naturally be the only way to build a radio receiver. But all was not rosy in superhet town. For one thing it was too expensive! A good triode cost about $6.50 per copy and, considering an 8-tube lineup, that would take a small bundle for tubes alone, taking into account what the dollar was worth in 1920. In fact, there were only a few components that were available for less than a $5 bill. Transformers, tuning coils, variable condensers, etc., all fell into this high-price bracket. Besides, the circuits were complex and difficulty arose from the fact that the oscillator was separately tuned because ganged tuning capacitors had not yet been invented. This allowed the operator to set the oscillator to a frequency on either side of the incoming signal which placed every station at two points on the oscillator dial. For example, the mixer tuned to 1000 kHz and the oscillator to either 900 kHz or 1100 kHz would give a difference of the required 100 kHz, so the station would show at both points.

Another problem reared its ugly head. The mixer was very broad in its tuning and powerful off-frequency locals could come pounding in even with the mixer set at 1000 kHz. If the operator had chosen the 1100 kHz spot for the oscillator and a powerful local was also broadcasting on 1200 kHz, it too could make its way into the mixer and produce the required 100 kHz which would really make a cat fight in the

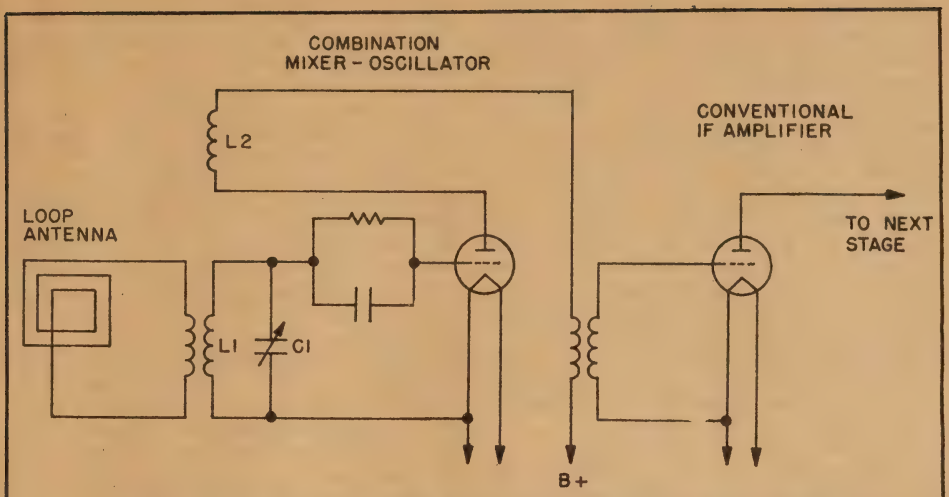


Fig. 2. Don't confuse this circuit with a regen receiver—actually, it is a superhet-type. Capacitor C1 is tuned to 100 kHz above the antenna signal and mixed in the first stage.

speaker. This latter problem, where the difference frequency is produced by a station you don't want, is called an image, and plagues low cost superhets more or less to this day. You might be able to teach a little old lady to work all the levers on her Stanley Steamer, but there would be no chance to expect her to puzzle out the oscillator tuning. Because of this and other problems, the ultra sensitive and selective superhet was temporarily left to the very experienced experimenter or radio amateur.

The TRF Had Its Day. With the skyrocketing popularity of radio broadcasting, the home entertainment center featured not the superhet but instead a tuned radio frequency receiver—TRF for short. Early radio magazine ads showed home life on a cold snowy night with a warm fireplace crackling and everyone seated comfortably around the TRF and listening attentively while Dad readjusted the three main tuning dials. Outside was a massive long wire antenna installation carefully rigged to hold up one or more strands of copper wire at a great height.

Regeneration. The radio usually was a 5-tube set with a tunable low cost RF amplifier. It was an instrument to behold by those just introduced to the electron art, but with all its squeals and whistles it left a lot to be desired. Armstrong was one of the first to recognize that in order

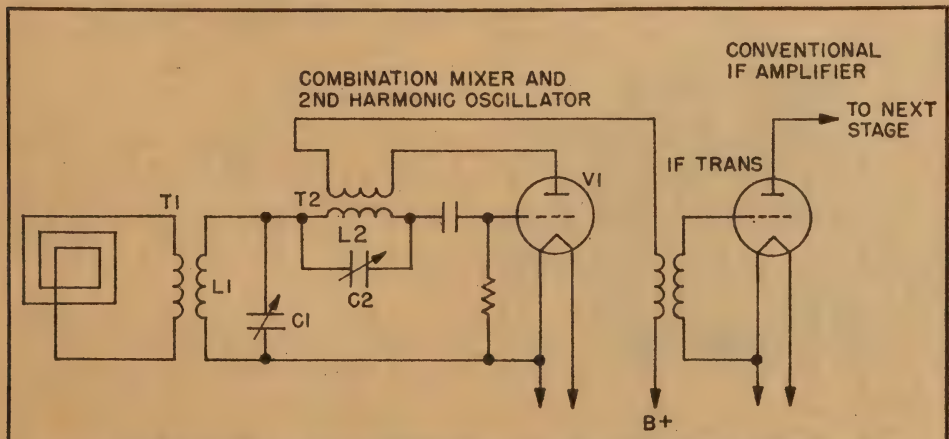
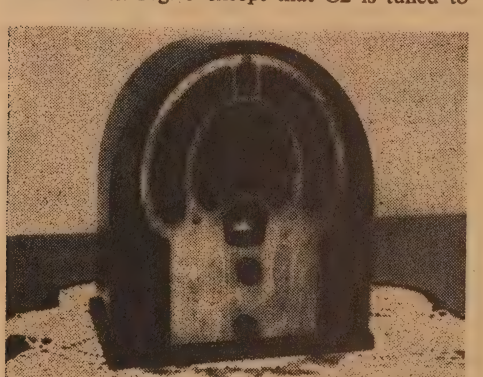


Fig. 3. Once you read the text explanation on how this circuit works, it will be difficult to understand that it works at all! Tuned circuits L1-C1 and L2-C2 interact, making it practically impossible to tune the receiver. Amstrong had to tune L2-C2 to half the desired mixing frequency to obtain the second harmonic for signal mixing.

for the superhet to make its appearance in the living rooms of America he had to reduce its cost and simplify its tuning. He began work, determining to reduce the number of tubes without chopping the performance. One way was to make the mixer tube also double as an oscillator. Consider the circuit shown in Fig. 2. Here the mixer stage is also made to oscillate by virtue of the plate signal fed back to the grid circuit through the inductive coupling of L1 and L2. If C1 is tuned to say 1100 kHz, then any 1000 kHz broadcasting signal that can struggle into broad tuning L1 will also be impressed on the grid and we have a resulting difference frequency of 100 kHz. It is then fed to the IF amplifier in the usual way. What's wrong with this scheme is that C1 is actually detuned from the desired 1000 kHz broadcast station so we lose sensitivity.

Next let's try the circuit shown in Fig. 3. Here the oscillator is set in motion by using the same general scheme of inductive plate to grid feed-back through T2. The radio station's modulated signal comes in on T1. Notice that L1 and L2 are connected in series. We can tune the L1-C1 combination to 1000 kHz and the L2-C2 combination to 1100 kHz. We then have in the plate circuit of V1, the 1000 kHz signal from the radio station plus the 1100 kHz feedback oscillator output which combine to give us our good friend the 100 kHz difference frequency. It sounds easy but it turns out that it was practically impossible to tune. In reality, the tuning of C1 and C2 are only 100 kHz apart and if you change the setting of C1 it interacts to alter the tuning of the L2 and C2 network and vice versa. So you would have to be an expert to keep up with that tuning marathon.

Armstrong's staff finally came up with the solution to the dilemma. They used the same circuit as in Fig. 3 except that C2 is tuned to produce an oscillator frequency exactly ½ of that used during the marathon bout. In other words, with C1 tuned to the usual 1000 kHz, C2 is tuned to 550 kHz instead of 1100 kHz. But why do that? Well, oscillator circuits in those days were rich in harmonics, so appearing at the IF transformer is not only the 1000 kHz component but also the second harmonic of the oscillator frequency, or 1100 kHz, which produces the nice 100 kHz needed for the IF amplifier. This placed the mixer tuning and oscillator tuning far apart in frequency by a factor of 100% and therefore practically no interaction exists between the two. So now we have a satisfactory mixer and oscillator in the same tube! This was a major receiver circuit breakthrough.



Every grandfather today can recall the Philco Model 80 design with its remarkable circuit innovations. Designed to sell during the Depression days, the Model 80 brought laughter and hope into the homes of Americans.

More Tricks. Armstrong wasn't satisfied with this. He needed to sharpen up the tuning of the mixer to eliminate the image frequency problem mentioned earlier. An RF amplifier ahead of the mixer was in order, but how to get it without adding another tube? He solved this by going ahead and adding the RF tube ahead of the mixer but also let it double as an IF amplifier. Armstrong was getting tricky.

Refer to Fig. 4 and we will see how he did it. The modulated signal comes in on the loop which is tuned to our usual 1000 kHz station by C1. This is amplified by RF amplifier V1 and is fed through L3 to the input of the 2nd IF transformer. (Note how V2 is bypassed.) Since the 2nd IF transformer is tuned to pass 100 kHz only, the 1000 kHz signal can't get through and stops there. Notice that while this was going on the RF energy is also passed inductively to L4 and mixes with the oscillator frequency by virtue of the same hookup shown in Fig. 3. C2 is tuned to 550 kHz and the components of the station signal and the 2nd harmonic of the oscillator show up in the plate circuit of V2. Obviously a 100 kHz difference is generated, which is routed back to the primary of the 1st IF transformer. It passes this frequency real well and impresses it back on the grid of V1. Now, V1 is looking at two signals, the modulated station frequency at 1000 kHz and the 100 kHz difference. Tube V1 amplifies both signals just fine and passes them on through L3 again and to the 2nd IF transformer which passes the 100 kHz component real well. The IF amplifier gladly amplifies it before passing it on to the following vacuum-tube stages.

So Armstrong now had another breakthrough. He had succeeded in incorporating all the virtues of a well designed superhet by using six battery-powered triode tubes. In conjunction with the packaging engineers of RCA, this circuit made its appearance in 1924 as the Radiola Superheterodyne. It sported a built-in loop antenna making the big long-wire antenna unnecessary. You might say this was the beginning of public acceptance of the superhet. It wasn't low cost ($286 without batteries); but thousands were sold because affluence was becoming

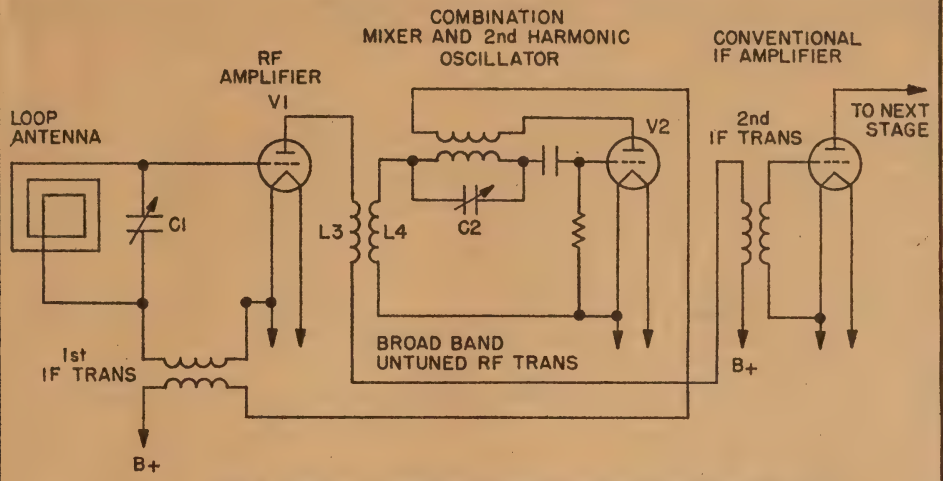


Fig. 4. Here is a circuit that defies analysis at first glance. The combination mixer and 2nd harmonic oscillator circuit operates like the circuit in Fig. 3. This receiver circuit was first used in 1924.

more common in the dawn of the great business boom of the mid 1920's. It was not too difficult to tune, although the separate oscillator tuning caused repeat points for every station. However, careful adherence to the instruction book reduced this. So now the little old lady with the Stanley Steamer could easily tune in these distant stations and have no difficulty in separating the Merry Old Chief of the Kansas City Night Hawks from Harry Reser's Clicquot Club Eskimos.

A New Dawn! The introduction of the Radiola Superheterodyne in 1924 breathed new life into the radio fan fraternity. The radio magazines of the day now featured a vast array of superhet circuits encompassing all sorts of theoretical advantages. Gerald Best, McMurdo Silver, Robert LaCault, and Jackson Pressley were some of the popular figures who offered design innovations. The public's hunger for selective reception of distance stations spawned bigger and better sets. Competition among radio manufacturers was bringing down the price of tubes and parts, so one didn't have to be quite as economically minded when cooking up a *super circuit*. Our friend the radio experimenter now bragged that his set tuned so sharply that when *tuning in a duet, it came through as a solo!* Superhet kits began making their appearance with easy step-by-step instructions for the not-so-learned radio fan.

With the advancing state of the art, the operating frequency of the IF amplifier started to increase. Instead of the popular 50 kHz and 100 kHz, in the early days, values of 175, 262, and 370 kHz made their appearance to help reduce image frequency. It is a well-known fact that 455 kHz is the popular one today. How did we ever settle on that value? This is kind of a mystery, but one reason already mentioned is to help reduce image frequency. For example, go back to the case of tuning the mixer to a 1000 kHz station, the oscillator would be tuned to 1455 kHz to produce a difference of 455 kHz. Now, if the mixer also allowed a 1910 kHz signal through, it would produce a 455 kHz difference and would come through on top of the 1000 kHz station. But it would be a pretty lousy mixer that would be so broad as to allow this to happen.

But there is a bit of folklore that had more to do with pegging 455, and it had something to do with harmonics. Notice that the second harmonic of 455 KHZ is 910 kHz. In the late 1920's there were no U.S. stations broadcasting on 910 KHZ. It was reserved for the Canadians. Legend has it that 455 was selected to protect the IF amplifier against interference. An improperly shielded 455 kHz IF amplifier would easily let a 910 kHz (2nd harmonic) station get through if you had a powerful one nearby. So, unless you lived on the border, this arrangement would help protect against that problem in your early superhet.

Bye-Bye TRF. With the further loosening of the economy, the general public began finding out about the superhet and the TRF began losing favor. In the late 20's the superheterodyne circuit flourished, with the appearance of the big $400 consoles which featured a single tuning dial and tremendously improved audio quality.

Since necessity is the mother of invention, when the bottom dropped out in 1929 the radio designer had to come up with something that could be had for a few bucks. Next came the entrance of the cathedral-shaped table radio.

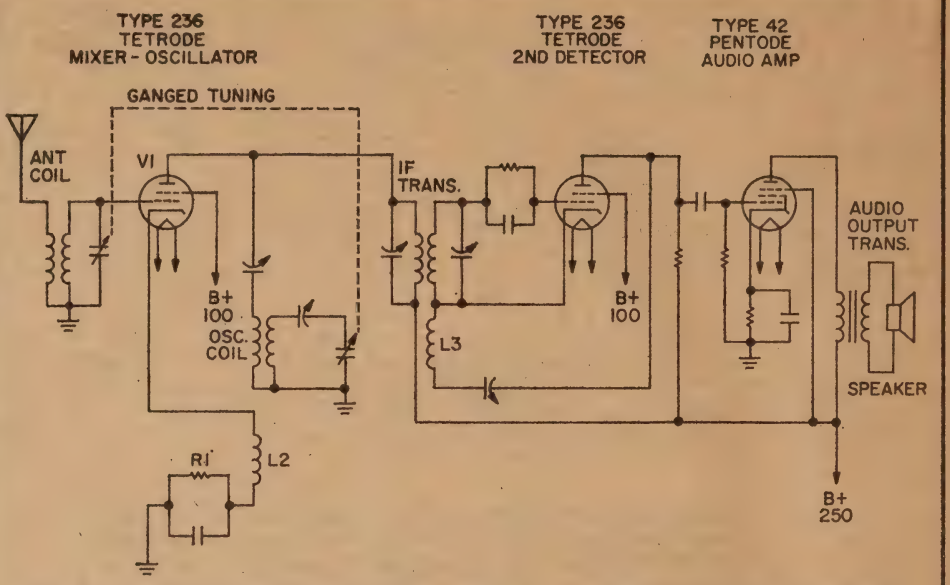


Fig. 5. Now the superheterodyne circuit is shaping up to resemble the circuit used today. One knob tuning was now a reality with the invention of the ganged tuning capacitor. This circuit was used in the 1931 Philco Model 80. The tetrode and pentode vaccum tubes were relatively new. A rectifier tube, not shown, provided DC voltage eliminating the need for expensive wet-cell batteries.

Radio designers now had to resort to all kinds of tricks in order to drastically reduce cost but still maintain good selectivity and tone quality.

One of the marvelous examples of such a play is the 1931 vintage Philco Model 80. All the superhet action is done with two tetrode tubes! Add a pentode audio stage and a power supply rectifier (4 tubes total) and you have a very fine set to sell for about $19.95. The circuit is shown in Fig. 5 and it operates something like this: connected to the plate of the 236 mixer-oscillator, V1, is an oscillator coil with its tank circuit tuned 455 kHz higher than the incoming signal. This energy is transferred by pickup coil L2 in series with the cathode bias resistor R1, so the plate current in V1 will vary at the oscillator rate. Also, the plate is varied by the incoming signal on the grid so we end up with the difference frequency of 455 kHz which is passed on to the IF transformer. They couldn't afford an IF amplifier tube, so they made up for it by introducing a little positive feed-back in the 2nd detector by virtue of coupling coil L3. Believe it or not, even though the actual circuit contains 8 trimmer capacitors which needed peaking up, the Philco Model 80 home receiver really performed with the one-dial tuning at that.

By now the little old lady with the Stanley had lost all her money in the stock market crash. She may be forced to drive a used Model T Ford, but she's fortunate in one way—she only needs a $20 bill to get a first class table model superhet that would eat alive her old Radiola Superheterodyne.

The radio development that finally sounded the death-knell to the TRF radio and made the superhet the front runner for good and all was the advent of the pentagrid converter tube. One of the first was the type 2A7 and was basically two tubes in one and allowed the mixing and oscillator action to happen in one envelope. All previous attempts described so far using conventional tubes depended upon the signal frequency and oscillator frequency being impressed on the same control grid. This arrangement loses efficiency as the oscillator frequency goes up because the signal and oscillator voltages get out of balance. The pentagrid tubes provided electronic shielding between the two functions and eliminated all the loading and detuning problems. The type 2A7 was the grandfather to the 6K8, 6SA7, 6BE6, etc., which were so popular in the twilight hours of tube-type superheterodyne radio.

History will certainly call Edwin Armstrong "Mr. Superheterodyne." His contributions were a major factor in leading the way to awaken the giant that we have come to recognize as the fabulous electronics industry. ■

Taking the Gamble Out of Gain

☐ Gamblers and amplifiers have one thing in common: Their goal is to get more *out* than is put *in*. But, while successful gambling requires the smiles of Lady Luck, amplification is not a matter which should be left to chance. All too often, however, amplifier circuits are used whose actual gain is a complete unknown until the circuit is actually built and tested.

And gain is not the whole story. The bias, or DC operating point, must also be known and predictable, or the amplifier may clip and distort the signal it is passing. In extreme cases, it will fail to pass a signal at all. Again, we often see amplifier circuits whose DC operating points are left to chance, or are "tweaked" into place only after the circuit is built and fails to operate correctly.

Such gambling with gain and bias is unnecessary. Very simple and straightforward methods can be used to predict the gain and DC operating conditions of an amplifier, making its behavior a matter of logic, not luck. These techniques succeed because they make the gain and DC operating points depend upon resistor values, which are predictable, instead of depending upon transistors and ICs, which are notoriously unpredictable, even at the present state of the art.

In the following paragraphs, we shall give examples of both *long shots* and *sure things* in the amplifier world, and show how to convert a long-shot risk into a much more reliable amplifier.

Russian Roulette Special. If you like to gamble, the circuit of Fig. 1 will give you a *lot* of gain—if it works at all! But even if you get it working, the first hot day may send it into raspy distortion as the hot transistor shifts its DC operating point, brutally destroying your clean hi-fi signals. And even if you can get it to keep its cool, you will find that the resistor values shown won't work if you change transistors, or if you try to build another identical circuit. In brief, this circuit requires, first, that you hand-tailor it to fit each and every transistor, and second, that you keep the circuit's transistor cool.

So in spite of the high gain available from this circuit, it represents a bad gamble—a real long shot—because it may not work at all, or it may work one day and not the next, or it may work with one transistor and not another. Even if Lady Luck smiles on you and the circuit works, you cannot readily predict how high the gain will be.

A Better Gamble. If you don't like long shots, you can add one resistor to the *Russian Roulette Special* and improve its reliability considerably. But, as in most gambling operations, the payoff—in this case, the gain—goes down.

Take a look at Fig. 2 and you will see that it is just the *Russian Roulette Special* with a resistor added in the emitter circuit.

The gain of this circuit is, for all practical purposes, completely independent of the transistor. In fact, the gain is simply the value of the collector resistor, R2, 10,000 ohms, divided by the value of the emitter resistor, 2,000 ohms. Therefore, the gain is 10,000/2,000 = 5. A one-volt signal applied to this circuit can be relied upon to produce a five-volt output signal, every time.

The new emitter resistor also does nice things for the DC operating point, by making it more predictable and less susceptible to high temperature difficulties. Although this circuit is still not the best we can do, it is a much better bet than the *Russian Roulette Special* shown in Fig. 1.

The Sure Thing. Adding yet another resistor, R4, to the circuit of Fig. 2 gives us Fig. 3, which is a real *sure thing* in the amplifier world.

Just as for the circuit of Fig. 2, the gain of this circuit is R2/R3. Since the value of R2 has to drop a little for this circuit—for reasons explained below—the gain of this circuit is 9,100/2,000 = 4.55. Note that, just as in Fig. 2, the transistor has nothing to do with this value of gain. Only the fixed resistors determine the gain, which is just the way we like it.

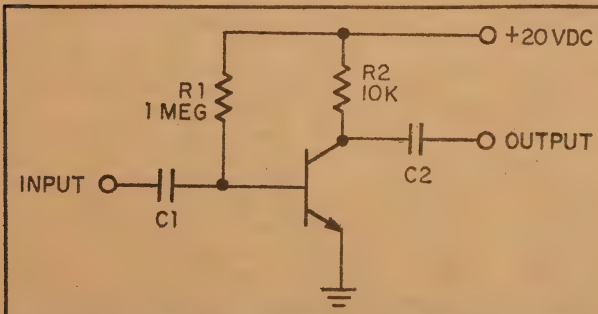


Fig. 1. This circuit is not too hot because even if you get it to keep its cool, replacing resistor R1 is necessary when you change the transistor. The Author calls this circuit the "Russian Roulette Special."

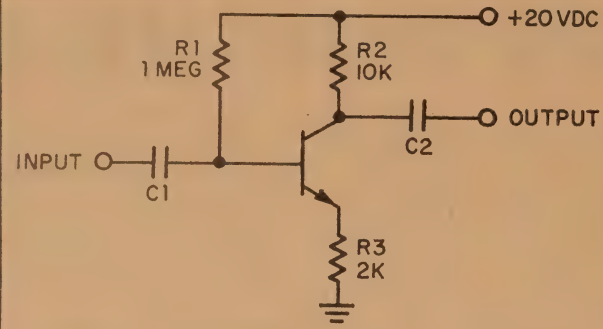


Fig. 2. This circuit amplies the input signal, and the amount of gain is not a function of the transistor, but the ratio of resistances R2 and R3. In this case, gain is 5.

The DC operating point of this circuit is extremely stable and predictable—far better, even, than the circuit of Fig. 2—and is determined by the resistor values only.

Putting It Together. If you're a compulsive gambler, or for some reason like to play *Russian Roulette* with your amplifier designs, you may opt for the simple but unreliable circuit of Fig. 1. In such a case, you can reduce your risk somewhat by following the step-by-step design arrangement given below:

1. Choose your power supply voltage. In Fig. 1, we chose 20 volts, a good round number. For most transistors, values below 2 volts will make for a difficult design. In no case should the voltage chosen be greater than the value given in the transistor data for BV_{ceo} (*B*reakdown *V*oltage; *C*ollector-to-*E*mitter, with the remaining electrode (base) *O*pen).

2. Second, choose the current desired in the transistor. For economy, especially for battery operation, this should be as small as possible. However, many small transistors do not operate well below 0.5 milliampere; often the transistor databook will indirectly recommend a current by saying something like "*Beta* (or h_{fe}) is 35 at I_c = 2 mA." This is a broad hint that the particular transistor type you're considering has been manufactured to work best at 2 mA. Under no circumstances should you choose the current value given under "Maximum Ratings." This is the red-line, never-exceed value of current; one increment beyond this value and the transistor may vanish in a puff of silicon smoke! If in doubt, 1 or 2 mA is a reasonable value for small transistors. In our example, we chose 1 mA.

3. Having chosen the current, calculate R2, the collector resistor, with the formula

$$R2 = \frac{500 \times E}{I_c\ (mA)}$$

$$R2 = \frac{500 \times 20}{1} = 10{,}000 \text{ ohms}$$

4. For the next steps, you need a voltmeter, an ohmmeter, a 2-megohm potentiometer, and a resistor somewhere between 10,000 and 100,000 ohms. Put the circuit together with the pot, resistor, and voltmeter as shown in Fig. 4.

5. Adjust the potentiometer in Fig. 4 until the voltmeter reads one-half the supply voltage—10 volts, in this case.

6. Without disturbing the potentiometer setting in Fig. 4, remove the pot and its associated resistor and measure their combined value with the ohmmeter.

7. Use the value obtained in step 6 for the R1 of Fig. 1.

To complete the design, C1 and C2 should

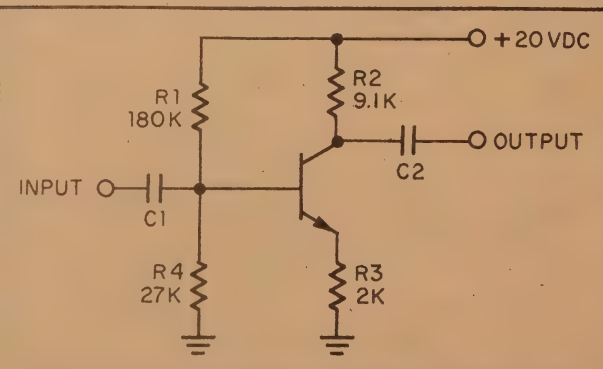


Fig. 3. Stability is the name of the game played with gain. The addition of R4 removes the circuit's tendency to self destruct like the one shown in Fig. 1. Gain is still the ratio of R2/R3, but notice value is lower, with a loss of some gain.

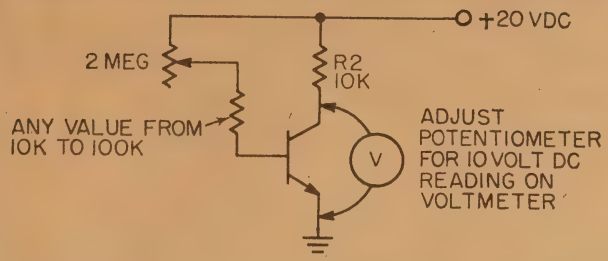


Fig. 4. Adjusting pot until voltage across transistor is one half of power supply is a sure way of discovering value of R1 in Fig. 1. Be sure potentiometer resistance is maximum and decrease slowly, never exceeding one-half collector voltage.

be chosen to have values in the 100 μF region for most audio work, with voltage ratings equal to or greater than the power supply voltage.

If you ever have to replace the transistor, you will probably have to repeat steps 4, 5, 6, and 7.

The above effort, which includes all that playing around with meters and pots, yields you an amplifier of unpredictable gain and doubtful reliability. The circuit of Fig. 2, though better, offers not enough real improvement to warrant a detailed discussion. Fig. 3 is the circuit that removes us from the ranks of risk-takers, so we should next see how this superior circuit may be designed, step-by-step.

No-Gamble Gain: Here's the step by step procedure that is foolproof.

A. Select the supply voltage and transistor current just as in steps 1 and 2, above.

B. Select a DC voltage, smaller than the supply voltage, which you want to appear at the emitter. In the example in Fig. 3, we chose 2 volts. If this voltage is chosen too small (less than 1/20th the supply voltage), gain predictability and bias predictability will be poor. If chosen too large (more than half the supply voltage), then so much of the available voltage is "used up" on the emitter that there's very little left to produce signal output. So, it's a trade-off between good stability and a large output. A good rule of thumb is to let about one-fifth to one-tenth of the supply voltage appear at the emitter.

C. Calculate the emitter resistor, R3, by the formula

$$R3 = \frac{1{,}000 \times E(\text{emitter})}{I_c\ (\text{mA})}$$

$$R3 = \frac{1{,}000 \times 2}{1} = 2{,}000 \text{ ohms}$$

D. Add 0.6 volt to the emitter voltage, to obtain the base voltage:

$$E(\text{base}) = E(\text{emitter}) + 0.6$$
$$= 2 + 0.6 = 2.6 \text{ volts}$$

(Note: Use 0.6 volt for silicon transistors; use 0.2 volt for germanium transistors)

E. Calculate R4 by the formula

$$R4 = \frac{10{,}000 \times E(\text{base})}{I_c\ \text{mA}}$$

$$R4 = \frac{10{,}000 \times 2.6}{1} = 26{,}000 \text{ ohms}$$

(use EIA value, 27,000 ohms)

F. Calculate R1 by the formula

$$R1 = \frac{10{,}000 \times [E(\text{supply}) - E(\text{base})]}{I_c\ \text{mA}}$$

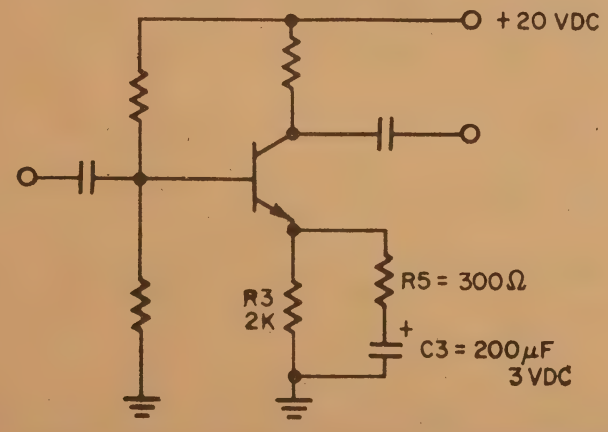


Fig. 5. With the gamble gone, go for more gain. With addition of R5 and C3, the circuit's gain is effectively increased to 34.9—that's an improvement over 4.55. R5 is effectively in parallel with R3 reducing the overall emitter resistance to only 261 ohms.

$$R1 = \frac{10{,}000 \times (20 - 2.6)}{1}$$

R1 = 174,000 ohms (Use EIA value, 180,000 ohms)

G. Calculate R2 by the formula

$$R2 = \frac{500 \times [E(\text{supply}) - E(\text{emitter})]}{I_e\ (\text{mA})}$$

$$R2 = \frac{500 \times (20 - 2)}{1}$$

R2 = 9,000 ohms (use EIA value, 9100 ohms).

Notice that using up two volts on the emitter caused us to have only 18 volts left for output. The mathematics then forces us to drop the 10,000-ohm collector resistor to 18/20ths of 10,000 ohms, which is 9,000 ohms. This causes some loss of gain (from a gain of 5 for the circuit of Fig. 2, to a gain of 4.55 for this circuit) which is the price paid for stabilizing the amplifier. Again, it's just like gambling—when the risk goes down, so does the return.

Can we hedge our bets? That is, can we keep the DC operating point stable, and still get more gain? Indeed, we can. The circuit of Fig. 5 shows how this is done:

In this circuit, a 300-ohm resistor, R5, has been added in parallel with the 2,000-ohm emitter resistor, R3, but with a capacitor C3 in series with the new resistor. Since the capacitor is an open circuit to DC, the DC stabilizing effect of R3 is unchanged—it's just as though R5 weren't there, as far as DC is concerned. However, for the signal, which is AC, the capacitor looks like a short circuit—it looks exactly as though R5 were soldered directly across the emitter resistor.

So, for the signal, the total resistance in the emitter circuit is effectively the parallel combination of R3 and R5:

$$R = \frac{R3 \times R5}{R3 + R5} = \frac{2{,}000 \times 300}{2{,}300}$$

R = 261 ohms

And, since the gain is given by R2/R3, the addition of R5 in parallel with R3 has changed the gain thus:

Before R5 and C3 were added:

Gain = (9,100)/(2,000) = 4.55

After R5 and C3 were added:

Gain = (9,100)/(261) = 34.9

The addition of R5, then, gave us nearly eight times as much gain, and did so without disturbing the excellent DC-operating-point stability of the basic circuit. What price do we pay for this improvement—that is, beyond the money cost of the resistor and capacitor? And, if dropping in a 300-ohm resistor yields eight times the gain, would a 30-ohm resistor give 80 times the gain?

Obviously, the smaller we make R5, the more gain we get, until finally, when R5 = 0, we get the most gain we can obtain from this transistor. At this point—with R5 shorted out, and the capacitor C3 therefore directly across the 2,000-ohm resistor, R3—the gain is identical to the gain obtained from the *Russian Roulette Special* circuit of Fig. 1. It is just as large, and just as unpredictable. Such a circuit, with R5 shorted out and C3 directly across R3, is sometimes used to obtain the most AC (signal) gain available from the transistor, *while still preserving the top-notch DC-operating-point stability of the Fig. 3 circuit.* Any transistor plugged into such a circuit will have the same DC operating point as any other transistor in that circuit, or one like it; however, the gain for the signal will differ markedly from one transistor to another.

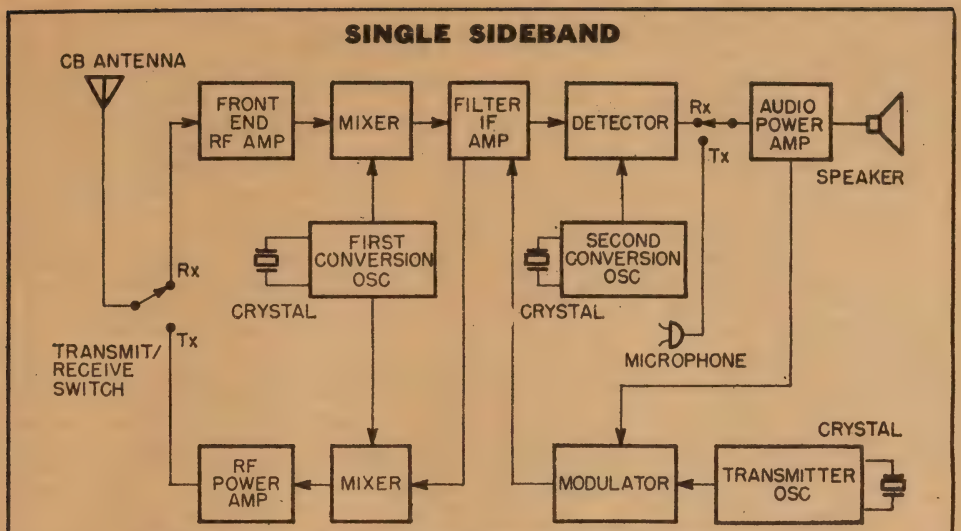
So gambling with gain and bias is completely unnecessary, since simple methods let us predict accurately the exact values of resistors needed to obtain optimum operation and reliability from our transistor amplifiers. Hook-up a few circuits today and checkout what you have learned. Don't gamble on your memory. ■

How CB 2-Way Radio Works

☐ The CB transceiver is truly an amazing device. Dollar for dollar it is the best value in consumer-type eletronic equipment. Organized around two basic design systems—amplitude modulation and single sideband—the modern CB transceiver can be specifically tailored to a given level of performance or operating convenience by adding a dollar or so of parts here, fifty cent worth of hardware there. No other consumer device can be so easily expanded to meet the need of the user. Whether you are a hobbyist, or business-only CB'er, that black box called a transceiver can do exactly what you want it to do when you want it.

The CB transceiver can provide general, or blanket, communications over a general area, provide pin-point line-of-sight communications, keep you informed of emergency situations, record messages while unattended, substitute for a telephone across inpenetrable mountains, function as a pocket-fone, and . . . well, anything you can think of in the way of personal communications can probably be handled by the modern CB transceiver. In short, the CB transceiver is a sort of universal "communicator" in that it can be made to meet any operating needs.

The exact facilities delivered by your present CB transceivers, or models you might be considering adding to your CB station, depends on



This block diagram of a modern single sideband transceiver clearly shows the additional circuitry required. To lower complexity,

the performance and features of small, individual "modules" which the manufacturer has plugged together at the factory. Using more or less basic circuits he can build upward from a three-control black box (channel selector, volume and squelch) to one of those gold plated specials with everything except running water.

In this article we're going to look into those "modules," for by having an understanding of how CB transceivers work, you'll be able to purchase a CB transceiver having *all the features and performance needed for all your applications.* As a side benefit, by knowing the in and outs of your CB transceiver, you'll be able to handle operating and minor repair problems which might come your way.

Two CB Systems. There are presently two distinct and separate type of transmission systems in use: AM, or amplitude modulation, and SSB, or single-sideband. They are incompatible with each other, meaning an AM station cannot "work" a sideband station and vice versa. Only if the individual transceivers have *both* AM and SSB can they work all CB stations.

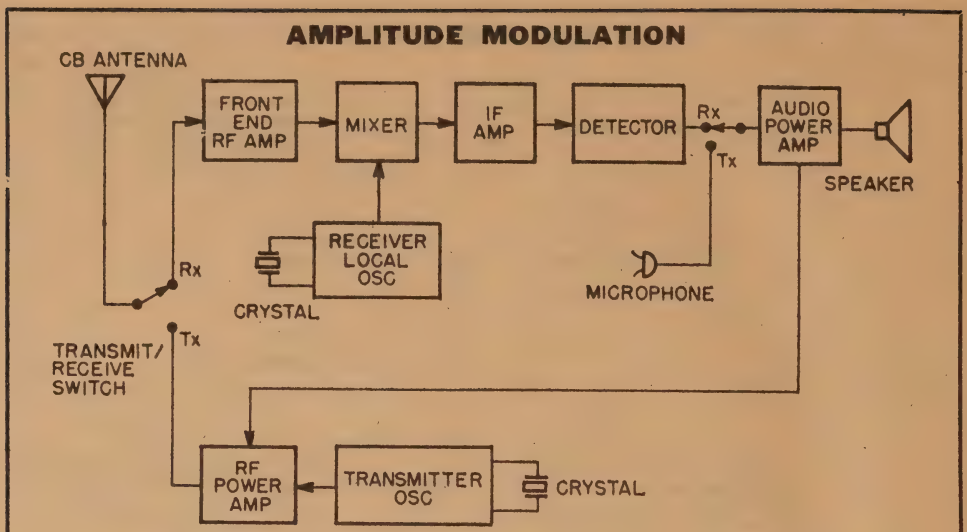
In the amplitude modulation system the voice energy, termed modulation, is impressed on the radio frequency carrier generated by the transmitter. In the process of combining the audio modulation and RF carrier, new frequencies called *sidebands* are generated. These sidebands are equal in frequency to: the RF carrier *plus* the modulating frequencies, and the RF carrier frequency *minus* the modulation frequencies.

The maximum amount of *undistorted* modulation that can be impressed on the RF carrier is one-half of the carrier power input, which, of course, also means one-half the carrier power output. If the CB tranmitter takes 5 watts input, the maximum amount of modulation is 2.5 watts. If the transmitter has 80% efficiency—actual output power divided by input power—there will be 4 watts carrier output power and 2 watts of modulation divided between two sidebands, so that each sideband has only 1 watt of modulation power.

One Does The Job. The AM receiver responds only to the energy in one sideband; it requires the carrier and other sideband for restoration of the signal back to audio, but only one sideband conveys intelligence. (The other sideband can be eliminated at the transmitter with very little audio loss at the receiver.) Therefore, of the entire 7.5 watts of energy put into the transmitter, at most, 1 watt of RF output power is converted back to an audio signal at the receiver.

The advantage of the AM system lies primarily in the fact that circuits are relatively simple, and it is quite easy to build a good communications system at a rather low price—as evidenced by some rather noteworthy CB transceivers in the $100 price range. It is the easiest system to use, requires no adjustments other than selecting the desired channel, and will generally give years of trouble-free operation even under the most severe handling.

Disadvantages to the AM system are excessive channel bandwidth and susceptibility to interference. Though only one sideband is needed to transmit intelligence, the AM signal has two sidebands, one sideband taking up space that could be used by another station. Though this is really no concern to the average user, the citizens band is fast becomming saturated with stations fighting for the available



Manufacturers double-up wherever possible by using the same circuit during both transmit and receive. SSB sets usually include the AM mode.

channels; if the second, unneeded, sideband were eliminated, the citizens band could accommodate twice the number of interference-free signals. As far as interference is concerned, an AM receiver is particularly susceptible to heterodyne interference, the whistle caused when two stations try to use the same frequency simultaneously. The interfering station's signal strength need be only 1/100 that of the desired station to cause objectionalable interference.

A Solution?—Single Sideband, or SSB as it is termed, utilizes the principle that only one sideband is required for transmitting intelligence. By eliminating the carrier and second sideband at the transmitter, all the intelligence-carrying energy goes into the entire signal transmitted by the station. This means that the transmitter input power allowed by the F.C.C., 5 watts average DC power, goes into the modulation energy. This is four times greater than the 1.25 watts of power input for one sideband of an AM signal.

To digress for a moment, you'll often find SSB transceivers rated at 10 watts P.E.P, or 15 watts P.E.P. This does not mean 10 or 15 watts power input. The P.E.P. stand for Peak Envelope power, which when integrated over the complete cycle comes out as 5 watts average power input, which is the same thing the F.C.C. means when they specify 5 watts DC power input.

All the RF energy radiated from the SSB transmitter is sideband energy, and since the power input of the sideband energy is four times that of the AM sideband input, the signal at the receiving station is four times stronger than that of an AM transmitter. Only problem is the receiver hears the sideband signal as monkey chatter or Donald Duck; completely, but completely unintelligible.

As we said, a carrier is necessary at the receiver for converting the signal back to audio. With a sideband receiver, the carrier is reinserted at the receiver by a separate oscillator which is part of the receiver's detector. (The detector doesn't care where the carrier comes from as long as one is there.)

23 times SSB=46. If we backtrack, we see SSB signal uses only the radio frequencies that are needed for one AM sideband—it could be the upper or lower sideband frequencies. This frees the frequencies needed by the unused AM sideband for another SSB station, and, in fact, two SSB stations can share the same channel without interference, one using the upper sideband frequencies, the other the lower sideband frequencies. If all CB stations were SSB the present 23 channels could provide 46 communication channels.

Single Sideband offers several advantages over AM. First, there's the matter of *working range*. The intelligence-carrying part of the signal is four times stronger than AM; added to signal enhancement by the process of SSB detection in the receiver, an SSB signal is the equivalent of 8 to 10 dB more effective than an AM signal. The effect at the receiver is as if an AM station were running almost 50 watts input power. Also, there is no carrier, so there is no heterodyne "whistle" interference if all stations using a channel are SSB. The interference between SSB signals sounds like chatter, which is not as annoying as a constant background whistle.

Since highly selective filter circuits are required for the generation of an SSB signal, these same filter circuits are sitting there, avail-

able for receiving, so you'll find that SSB transceivers feature high selectivity (adjacent channel interference rejection).

Since SSB requires considerable precision circuitry, as you'd expect, SSB rigs are not cheap. Unlike AM transceivers, which can be stripped of circuits to the bare bone and still deliver excellent overall performance, SSB requires sophisticated circuits for both budget and higher priced models—the differences in price generally reflected operating convenience features rather than performance.

The figure gives a good illustration of the difference; "A" is a basic AM mobile transceiver; "B" is a basic mobile SSB transceiver. If we consider that each individual circuit function is a *module* or *building block*, you can see that many more modules or building blocks are required for SSB. While the simplified AM transceiver can be produced as a reliable, low cost unit suitable for mobile operation, the basic SSB transceiver requires considerable circuit refinements and precision for fuss-free operation. The extra circuitry required for SSB is reflected in its substantially higher price compared to AM CB transceivers; however, because SSB transceivers must be built with a high degree of precision, all SSB models are inherently "gold plated specials" offering the highest level of performance.

Preview. For the remainder of this series we'll look into the individual modules that make up a CB transceiver, and show you how each module directly determines how much performance you'll get in your particular CB application. Next issue we'll take a look into the receiver, and see why one transceiver picks up two or three channels simultaneously, while another transceiver gets only the signal you want to hear. ■

All About CB Antennas

☐ Everyone in CB starts out with approximately five watts of transmitter power—actually 2 to 4 watts output but who's quibbling. But after the power leaves the set, the differences become enormous. Your signal may reach out ten miles, while a neighbor easily communicates forty miles. Results of that same power in the hands of NASA engineers can be phenomenal; space men can take five watts and relay data from a craft circling Venus 22 million miles away!

The difference, of course, is in the transmitted frequency and antenna system. It's the antenna that is the single most influential item and the one you can do something about. Proper antenna selection, installation and operation explain why some people put out a pounding signal, while others drown hopelessly in the noise. Maybe you can't construct a multimillion-dollar receiving horn (NASA's secret for getting 4-million miles per watt), but there are things you can do to improve range. A good place to start is with some info on how these CB sky hooks operate.

The Dynamic Duo. Two members of radio's nobility, Hertz and Marconi, lend their names to principles concealed in today's antennas. The Hertz antenna, or *half-wave dipole*, unlocks the theory behind almost all others. If you think of radio signals as a series of water waves—a comparison, incidentally, that makes professors wince—you can get a half-decent view of the half-wave at work. One complete radio-wave cycle is shown in Fig. 1A. As it emerges from the transmitter it travels down the coax in a succession of waves that measure 36 feet in length for CB. Instead of crests and troughs of water, each wave consists of electromagnetic energy which constantly reverses its polarity—the number of polarity changes in each second gives us the frequency.

Half a Dipole. Consider a basic antenna for sending or receiving such a wave. It's the *half-wave* dipole, or one that's cut to 18 feet for CB. As we'll see, it's all that is needed to accommodate the complete 36-foot signal since only one polarity of the wave is present at any given time. Refer

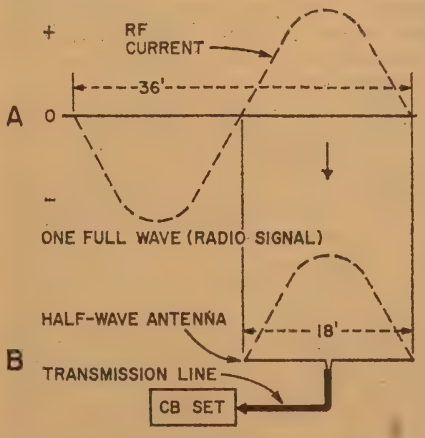


Fig. 1. RF current distribution in half-wave antenna at maximum positive peak sine-wave.

to Fig. 1B. Note that the antenna is capturing the positive portion of the signal in our example. An instant later, the wave will change polarity—to its negative phase—and excite the wire with current of the opposite polarity. Thus each half of the wave takes its turn occupying the wire.

The half-wave dipole is not only an efficient antenna for its length, but it's a standard against which the engineers measure the performance of all others. It is not a practical CB antenna because better performance is possible by special designs which evolve from the half-wave idea.

For cars, a half-wave CB antenna would be mechanical folly; an 18-foot rod would take a real beating from trees, power lines and underpasses. Fortunately, it's possible to cut a half-wave antenna in half to shrink its proportions to reasonable size. The result, shown in Fig. 2, is the basic Marconi

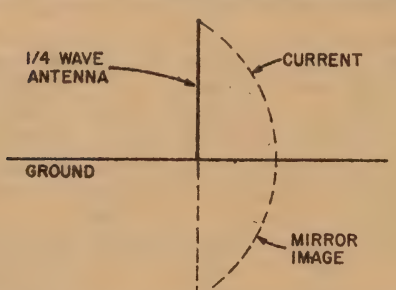


Fig. 2. ¼-wave antenna is popular for mobile. May be physically shorter with loading coil.

antenna, known now as the quarter-wave whip. The ground portion of the system simulates the missing half of the original antenna (note the mirror image) and the radio wave gets the same treatment as before. Now a mobile antenna can be half the height—9 feet—and still be tuned to the signal.

Technical Considerations. Before we see how manufacturers take these basic antennas and come up with dozens of variations, consider some restrictions. Although radio waves of lower frequency hug the earth and travel around its curve, the 27-MHz CB variety are propagated mainly line-of-sight. When sunspots are numerous, there is much long-range skip as signals bounce back to earth. Sunspots are now diminishing and the mid-70's should create poorer hunting grounds for illegal skip-workers. But the horizon, hills and other obstructions restrict normal range unless the antenna is placed at the highest possible location within the legal limit. This is basically 20 feet over a man-made or natural formation.

Radiation angle is another quality. Antennas vary in their ability to launch a signal that remains close to the earth's surface, as illustrated in Fig. 3A. Note that a high angle of fire causes loss of coverage—so the lower the angle, the better.

Polarization tells whether the radio wave is transmitted along a horizontal or vertical path. Usually this is determined by how the antenna is mounted: in the horizontal or vertical plane. The standard for CB operation is vertical polarization because it's not only easy to mount a vertical whip on a car, but the antenna radiates in every direction. This is important for mobile operation since the base station may not know where the car is located, although there are CB communications on certain horizontally polarized and directional CB antennas, as described in a moment. With this basic theory we can analyze major CB antenna types now being offered, starting with the base station.

An Old Timer. Once a popular antenna, the *ground plane* is waning because improved models can put more metal in the air where it counts. The ground-plane occasionally appears as an economy model because it's simple to construct and the cost is low. It is usually a 9-foot vertical whip similar to the quarter-wave antenna described earlier. But when placed atop a building, there's no metal car roof to supply the missing ground, so the ground-plane antenna must have *radials*. These are three or four horizontal rods which droop out of the base of the antenna. Though widely spaced, they electrically perform as though solid ground surrounded the antenna at the bottom. Effectively, they supply that mirror image aloft, where it's needed.

The Big Stick. An antenna which outwardly resembles the ground-plane, but has features of its own, is the half-wave vertical shown in Fig. 3B. Instead of 9 feet tall, it's double that height and thus presents greater area for signal radiation. The radials stemming out from the bottom contain *loading coils* which provide the electrical effect of much longer rods. One item not visible in Fig. 3 is a matching coil inside the fiberglass base. It's there for a good reason. Recall that the basic half-wave antenna shown earlier had a transmission line connected at the *center* of the dipole. The reason is that the

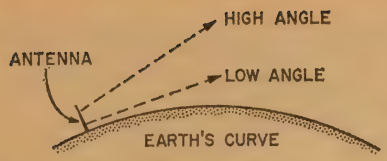
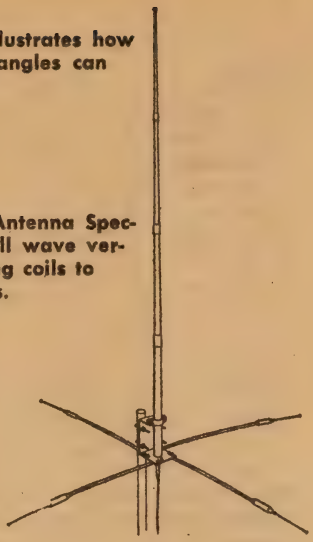


Fig. 3A., left, illustrates how high radiation angles can waste power.

Fig. 3B, right, Antenna Specialists M-317 full wave vertical uses loading coils to shorten radiaals.



half-wave antenna at this point displays an *impedance*, or electrical load, very close in value to that of the transmission line. The standard line for CB is 52-ohm coaxial cable and such a line may connect directly to the center of the half-wave antenna for maximum transfer of energy. But try to connect coaxial cable to the *end* of a half-wave antenna and a tremendous mismatch wastes almost all the power. So the manufacturer installs a matching coil at the base to transform impedance and come up with a proper electrical match.

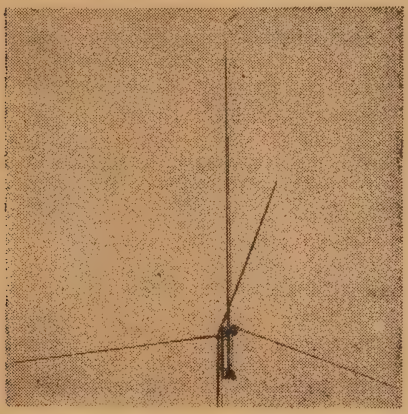


Fig. 4. Another version of groundplane, ⅝ wave provides gain. By Hy-Gain, the CLR2.

Another model is the 5/8-wave antenna shown in Fig. 4. It reaches up to just under the maximum height of 20 feet and uses radials for an electrical ground. (The radials also prevent the radiated power from being wasted in the cable coming up from below.) This model, incidentally, uses the designation "colinear," which means it electrically operates with elements stacked in the vertical direction. The antenna, however, incorporates coils which shorten the total height to keep it within the regulations.

A variation in base antennas is the one shown in Fig. 5. Most antennas tend to radiate strongest signals near the base where the cable is attached. This is where maximum electrical currents usually exist and it's current that's converted into the radio wave. In this antenna, the designers raise the radiating point toward the top. They state the higher position raises the signal so it's better able to clear nearby obstacles.

Power Booster? Search through antenna literature and you'll see statements that promise to "convert your 5-watt signal into 40 watts." How's it possible without violating the 5-watt law? The answer is that radio signals are squeezed into a narrow arc instead of radiating in every direction at once. The idea is not unlike a bare light bulb whose illumination is focussed by lenses or reflectors into the slim beam of a spotlight. The concentrated light is many times brighter—but only over a small area.

The same effect is easy to achieve with radio signals by adding elements which are slightly longer and shorter than the basic

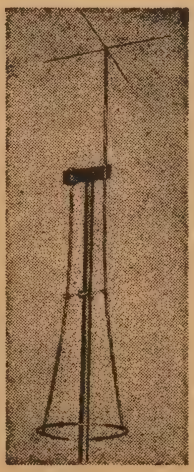


Fig. 5. A new twist in CB antennas, this vertical radiator pumps power from near its top to skim nearby obstacles. From Avanti, its the Astro-Plane.

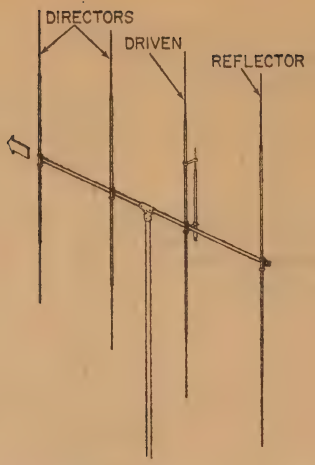


Fig. 6 Additional elements direct and reflect to provide greater Effective Radiated Power.

half-wave dipole. For example, look at the five-element beam in Fig. 6, which is aiming toward the left. The Driven element, which is a half-wave dipole, receives the signal from the transmitter and emits it to the right *and* left. The part of the signal that strikes the Reflector is reflected back in the desired direction (leftward). The first Director also intercepts part of the signal from the Driven element, only this time the signal is radiated again, but in the desired direction. The remaining Director does the same thing as its neighbor. The secret of the elements ability to reflect or direct is in their length and spacing. These factors cause the wave to be delayed (or phased) just the correct amount to cause reinforcement or cancellation in the desired directions. The result is a powerful "lobe" of concentrated wattage.

A beam can only have one reflector, but the number of directors vary. (The 5-element beam, for example, has three directors.) In the example shown, the manufacturer states a 9.4 dB gain, which means the 5-watt signal has the effect of about 20 watts. Highest gain occurs when two complete beams are "stacked together." You must, of course, use a rotator motor with most beams to direct the lobe of energy to the desired direction. Little signal is sent and received from the sides or back of the array. Some CBers use two antennas for this reason: a non-directional type to first acquire the distant station, followed by a switch to a beam for better communications. Note in Fig. 7 that loading coils are used in a 3-element beam to reduce its overall size to about two-thirds that of a conventional beam.

A beam which needs no rotator is the three-element "scanner" shown in Fig. 8. The CBer has a control box next to his rig to switch in pairs of elements which determine the desired direction. One position is "omni" to energize all three rods for conventional reception and transmission in all directions (omnidirectional).

Several other beam antennas include one model that contains both horizontal and vertical elements (Fig. 9). The CBer has a control box that selects either set for choosing the direction of polarization. By selecting horizontal polarization, for ex-

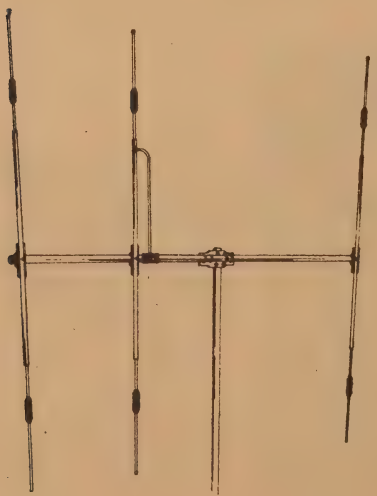


Fig. 7., left, Mosley GA-3D Mini-beam uses loading coils to simulate longer elements.

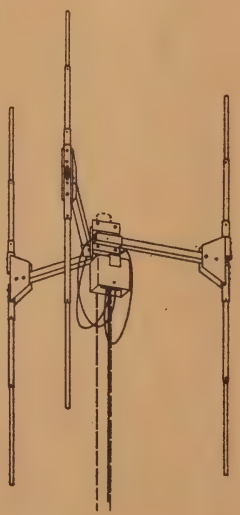


Fig. 8., right, Antenna Specialists M-219 has electronic switching for non-mechanical beam rotation.

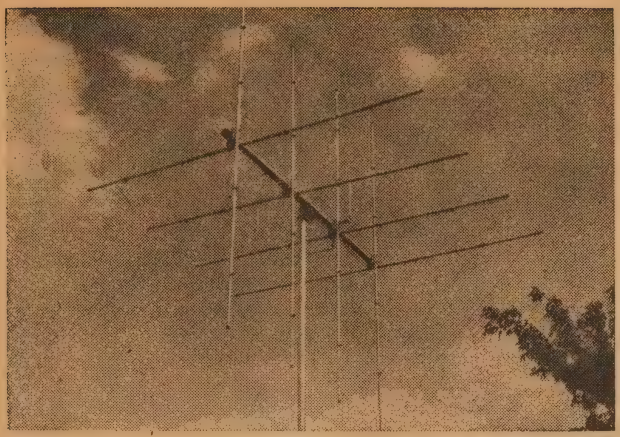


Fig. 9. Multi-polarized beam, Cush Craft CFB-8 lets CBer select best polarization mode to fit conditions, left.

ample, he might reduce interference from signals transmitted in the vertical mode. He might obtain better communications with another beam that can also select the polarization to produce strongest signals. The cubical quad antenna shown in Fig. 10 is another beam-type whose mode is switch-selectable.

The Mobiles. Antennas for the automobile do not have the variety of shapes enjoyed by their brothers on base, but they show plenty of ingenuity. The reason is that mobile signals would best be transmitted and received from a 9-foot rod growing out of the center of the car roof. Since this is thoroughly impractical, a full-size quarter-wave whip ends up on the bumper. There it may suffer from low position and distortion in its pattern; the signal tends to travel best in the direction of most metal. For example, a whip on the rear right bumper operates strongest toward the front left. This has led to a variety of shortened whips for mounting in more favorable positions.

Shrinking an antenna down from the 9-foot size is done by the loading coil, but the position of the coil is subject to much variation. A low point is good for mechanical stability and protection. A high point, on the other hand, is good because it lifts the signal further above the roof—but it's more susceptible to damage. Another problem with a lofty coil is that it is electrically sensitive to any swaying in the antenna. It could cause fluttery signals by detuning the rod. A compromise chosen by some manufacturers is the *continuously loaded* whip. The whole antenna is a coil—a wire embedded in plastic which spirals all the way up.

Which way to go?

One controversy that's hard to settle, though, is what's better, a small antenna (say 2½ feet) with a large loading coil placed atop the roof, or a 5-foot whip with a smaller coil placed lower on the trunk deck? Both antennas are electrically similar, but a larger loading coil tends to waste slightly more energy than the longer rod. Yet the smaller antenna permits a higher mounting point. Consider, also, that a roof-top mounting involves a more lengthy installation since you'll have to fish a coaxial cable through the car headliner. And so the dilemma continues. . . .

On the problem of antenna mounting the news is good. You'll find a tremendous variety of mounting styles. There are mobile antennas which grip the car through magnetic pull, clip on in moments, insert into the trunk-lid crack with no holes, while

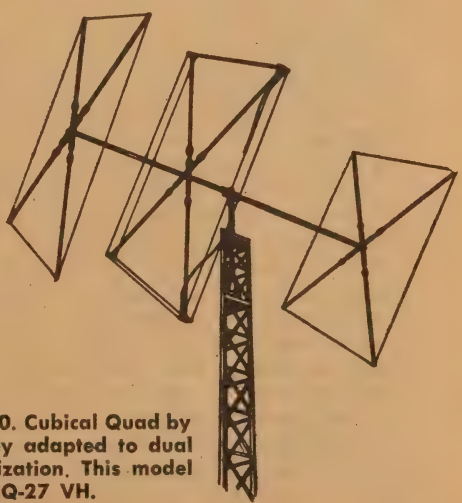


Fig. 10. Cubical Quad by Mosley adapted to dual polarization. This model is MCQ-27 VH.

others neatly fasten to campers or trucks. As our photos show, these and other mounting styles offer something for everyone. There's even a double-whip arrangement which derives a beam-like effect on an automobile.

A Pattern Emerges. In preparing this article, I asked several major CB antenna manufacturers the same question: "Where does the CBer go wrong in his antenna installation?" The most frequent cause of trouble isn't the antenna, but the coaxial cable. The line develops short circuits through poor soldering technique at the connectors, or sometimes the cable is crushed during installation and its electrical value changes. If soldering isn't your dish, purchase one of the prewired cable assemblies now widely available. Solderless coaxial connectors also ease the job.

One complaint echoed by some manufacturers is that the new CBer is usually surrounded by buddies with helpful advice on antenna installation. Bitter experience suggests something's amiss. If all manufacturers could sing one line of good advice, the lyrics would surely be: *Read the instructions!* ■

Getting CB Signals Out

☐ An old timer in the radio game once cleared up the mystery of CB antenna matching this way: "Imagine driving a car on an icy road. You gun the accelerator but the rear wheels spin uselessly and you don't go anywhere. Throw sand under the wheels and away you go. What you've done is matched power to load—just like matching a CB rig to an antenna."

There may be little connection between Edsels and electrons, but there's truth to the old timer's analogy. His spinning wheels represent the final radio-frequency amplifier acting as a power generator. Ice represents the load, or CB antenna, while sand is equivalent to a matching device which enables the wheels to bite into the ice—much like an amplifier delivering watts to an antenna. When matching is poor in either case (rig or car) the result is often useless heat.

The need for CB antenna matching springs from the differing impedance of various elements along the line. Measured in ohms, impedance is the AC resistance encountered by the radio signal in going from RF amplifier to transmission line to antenna, and finally to the air. Each element harbors its own characteristic impedance and maximum power transfer won't occur unless all impedances agree—which is to say "matched." What determines impedance value is often the internal nature of the device. Fortunately, it's possible to tamper with impedances and electronically juggle the numbers. As a rule-of-thumb, a transmitter stage that operates at fairly high voltage and low current displays high impedance. Thus, the final stage of a tube-type CB rig is generally considered high impedance because plate voltage is about 250 and current flow only about 20 milliamperes. These conditions help drive impedance up to thousands of ohms. The impedance of a transistor output, on the other voltage is typically low and current flow high. These shifting possibilities challenge the CB designer, who must build his creations to feed a 50-ohm impedance. Why that figure?

Picks a Number. Fifty ohms is a standard value which grew out of fifty-odd years of two-way mobile radio. It's handy for several reasons. One is that a transmission line (connecting rig to antenna) can operate at low impedance, which gives it low operating voltage. The line can safely snake up to a roof or through the car trunk. Also, a standard mobile whip has an impedance of about 50 ohms and there's less of a matching consideration.

As you can see in Fig. 1, the main points for antenna matching start at the RF amplifier final. In this circuit there's a matching device between the output stage and the 50-ohm coaxial cable. The "matcher" is supplied inside the rig by the manufacturer and is usually adjustable by the buyer, but more on this in a moment. The circuit transforms high tube impedance down to the low value of the cable. The second "matcher" is shown as an optional component because, in many instances, coaxial cable hooks directly to the antenna. (Antenna makers also adhere to that 50-ohm value for input impedance of a CB antenna.) An optional matcher, however, could be used between the line and antenna. Some antennas (beams, for example) may need

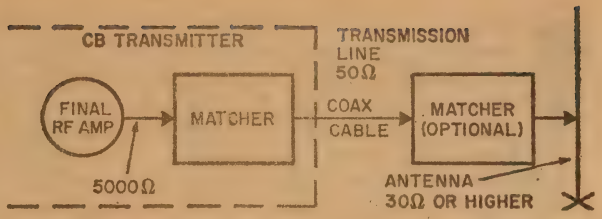


Figure 1. Matcher transforms impedance of final RF amp to that of antenna. Optional matcher lowers Standing Wave Ratio (SWR) of voltage within coax cable. Tune both matchers for lowest SWR reading with field strenght meter.

tuning, and accessory antenna matchers are sold for those who want to squeeze the last bit of energy out of the cable running to the antenna.

Got a Match. Unless these major points are matched, maximum power cannot flow. For example, if a high impedance source directly feeds a low impedance load, there'll be a loss because the load acts to short-circuit the energy. If the condition is reversed—low to high connection—then the load can't absorb full power. Further, whenever a bad mismatch occurs, the system suffers "standing waves." These demons develop if power fails to be soaked up by the load. Signals reflect back to the transmitter and cancel part of the power.

Bad mismatch in a solid-state rig can also blow the final transistors. This often happens as excessive voltages appear at transistor collectors and puncture the semiconductor material. Only through correct antenna matching will the 5-watt CB signal —weak enough to begin with—reach out for maximum range.

Have Some Pi. How designers accomplish matching from final amp to the line in a typical rig is shown in Fig. 2. It's the popular pi network formed by two variable capacitors around the final tuning coil. It permits an adjustable match between the tube and a reasonably wide range of impedances in the line. Impedance-transforming action occurs when the capacitors are varied and the output checked on a meter. You tune each control for maximum, being sure to repeat the adjustments several times back and forth to find the comination that produces best power output.

A similar circuit, found in transistorized circuits, is shown in Fig. 3; a double-pi network. The difference is that a tuning capacitor is joined by two coils with tunable slugs. The combination of components not only resonates the final amplifier to the desired output frequency (27 MHz) but transforms final transistor impedance down to that of the transmission line. A double-pi is used because it provides good opposition to the passage of second-harmonic signals from the final stage. These harmonics fall on 54 MHz and could cause considerable interference to TV channel 2.

In many instances, that's all the matching circuitry you'll find in a CB antenna system. Since line and antenna are each 50 ohms, they connect directly together with no further matching. If the transmitter output adjustments are correctly done, the rig should be driving the antenna with maximum energy.

The Meter Dead-out. But are the matching controls properly adjusted? Manufacturers usually preset the rig's output controls for a 50-ohm load at the factory, but this might not be the precise value for your installation. The position of an antenna and cable may affect impedance so to be sure antenna adjustments are satisfactory, you'll need some touching up. Some rigs have output indicators (Fig. 4) which tell if the final is putting out maximum signal, but they read indirectly and could be misleading.

A field strength meter (Fig. 5) can more accurately reveal the effect of transmitter adjustments. It's actually a primitive receiver that gives steady indication of carrier strength. Try to place the meter as far as

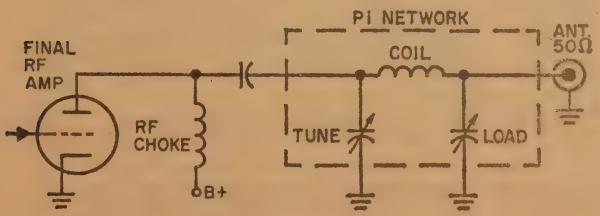


Figure 2. Schematic of pi-type output impedance matcher found in tubed CB transceivers. Network has two jobs; matching antenna to final RF amp, reducing second-harmonic content in output signal. Adjust load capacitor for lowest SWR.

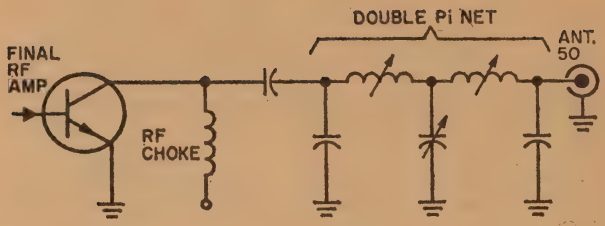


Figure 3. Schematic of pi-type output impedance matcher found in solid state CB transceiver. As with network found in Fig. 2, double pi network not only lowers impedance of output signal, it also resonates final amp to output frequency on CB band.

possible from the antenna while taking readings since it's possible to sense magnetic radiation from the antenna, and this energy doesn't contribute much to range. With the rig on, and the mike button keyed, tune the output controls (Fig. 6) for highest field strength. If you have a solid-state rig, don't turn any control more than a turn or so if there's no change in output. You don't want to risk a severe mismatch and possible damage to the semiconductors.

The field-strength meter suffices for tune-up, but an SWR (standing-wave ratio) meter will do an even better job. Its operation is based on the condition that when all components of the system are working at highest efficiency, SWR is lowest. The meter (Fig. 7) is inserted in the line and adjustments made for least SWR reading. When making these adjustments it's a good idea to also monitor relative output on the same meter (either reading is possible). You might mistune the controls enough to knock down RF output and believe efficiency is fine because of low SWR. A constant check on relative output prevents this error.

The SWR meter is also a handy trouble-shooter. If you develop a short circuit in an antenna element or transmission line, the SWR reading will shoot up and spell trouble.

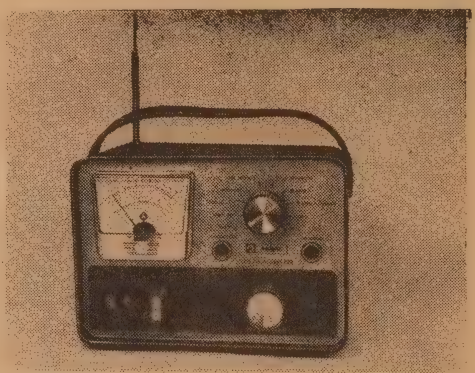


Figure 5. This mini test lab checks health of entire CB rig from mike connector to antenna. Shown is Knight's Ten-2 CB Checker.

A perfect SWR ration is 1-to-1, but this is impossible in practical circuits. Anything below, say, 2-to-1 is acceptable and means that little power is wasted in the system.

Before the Antenna. There are a number of accessory antenna matchers on the market and they can be installed by anyone who wants to squeeze the last milliwatt out of his rig (Fig. 8). They are similar to the matching circuits inside the CB rig; usually consisting of tuning capacitors and coils. The difference is that you can install a

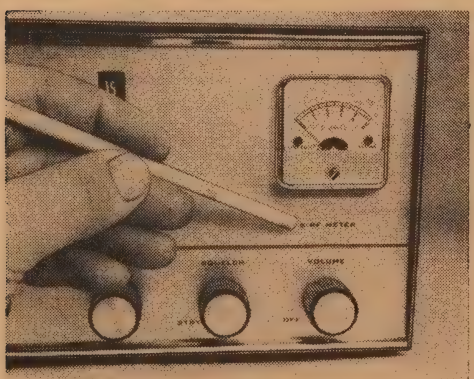


Figure 4. Indirectly-read S/RF meter can't tell you output SWR. Tune up this kind of rig with aid of field strength meter (FSM).

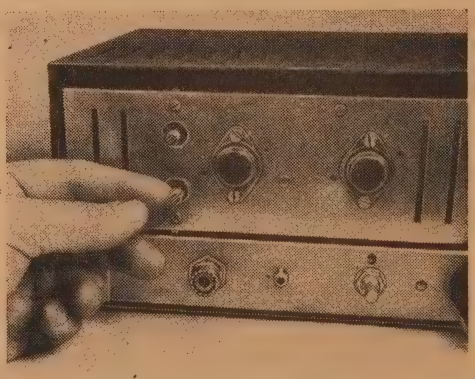


Figure 6. With your rig on, mike button depressed, adjust output loading, tuning controls for highest field, strength reading.

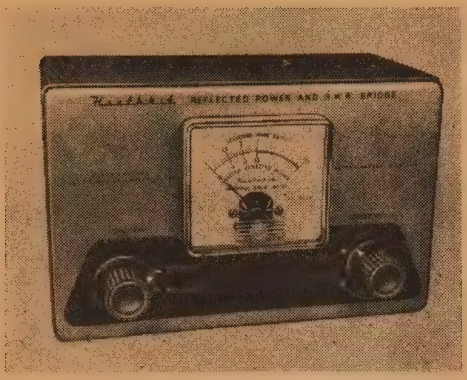


Figure 7. Keep tabs on your CB rig with Heathkit's Model AM-2 Reflected Power and SWR Bridge. Hams also find it useful.

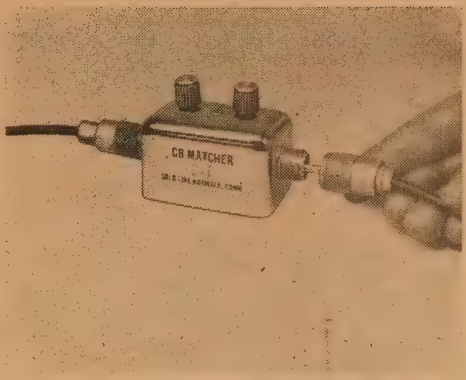


Figure 8. Think you can wring more power from your rig with an optional matcher? Here's one made by Gold Line.

matcher at the point where the line connects to the antenna.

Inserting a matcher at the output of a transmitter shouldn't be necessary because matching adjustments are already provided by the manufacturer. A good spot for these external matchers is at the base of a mobile antenna (in the trunk, for example). The reason is that mobile whips are rated at 50 ohms, but the car's shape and the height of the antenna may alter their value. In general, a whip suffers a mismatch because its impedance drops to below that of the coaxial cable when mounted on a car. A matcher can take care of this discrepancy by transforming impedances between line and whip until they agree. This is done by turning the matcher controls while observing either the RF output for maximum of for least SWR indication, as already described.

Coaxial Cable. There's a bit of controversy surrounding coaxial cable and how it's matched to the rest of the system. If you purchase the most popular cable—RG/58U—the manufacturer guarantees about 50 ohms impedance. This is implicit in the cable's construction; its center wire and shield act electrically as a series of tiny distributed coils and capacitors. No matter where you cut the cable, it presents a 50-ohm value (at either end). For this reason, a transmission line doesn't have to be tuned or matched, but merely cut to any desired physical length.

Here's where the hassle begins. The cable is 50 ohms—but only if you connect it to 50-ohm devices at both ends. Unless your transmitter and antenna are each operating at 50 ohms, the cable is knocked for an electrical loop and runs at some other impedance. Standing waves created along the mismatched line rock those little coils and capacitors formed by the coaxial conductors. These problems should be cured when you tune the transmitter output controls, adjust the antenna (if possible in your model) or install an antenna-matching box.

Some authorities claim, however, that cutting the coaxial cable to a specific length cures some of the loss from cable mismatch in a system that's less than perfect. The cure is based on the fact that a coaxial ca-

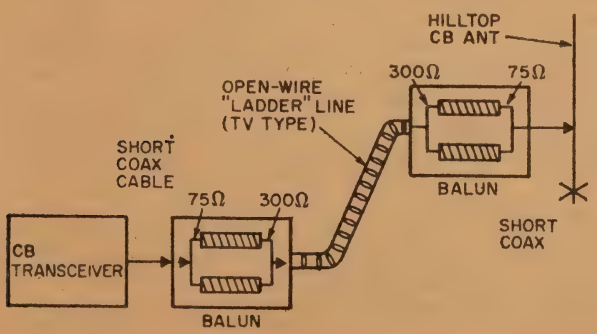


Figure 9. Open wire lead-in offers CBer one distinct advantage. Signal losses with this kind of lead-in are very low—hundreds of feet of cable can snake between rig, antenna without CBer encountering extreme signal loss. Only problem with "ladder" line is problem of impedance matching; easily solved with balun coils terminating cable at both ends.

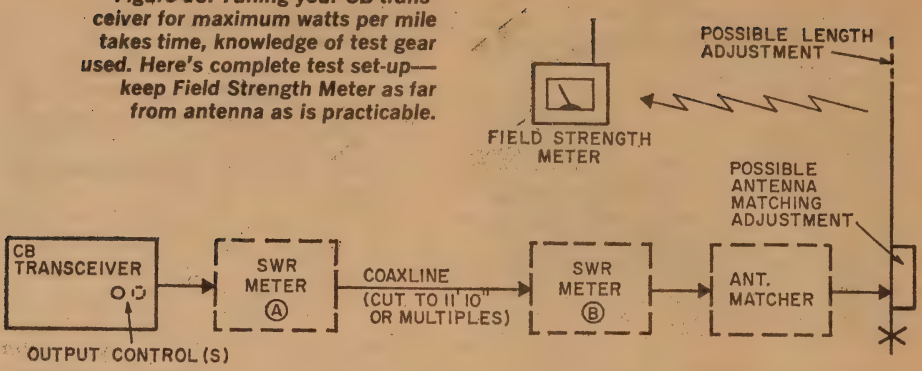


Figure 10. Tuning your CB transceiver for maximum watts per mile takes time, knowledge of test gear used. Here's complete test set-up—keep Field Strength Meter as far from antenna as is practicable.

ble can function as an impedance-matching device when cut to an electrical half-wave length of the signal. To turn this into practical numbers, you'd cut the cable to 11 feet 10 inches—or any multiple of that figure if a longer run is required. The next two length possibilities, for example, would be 23 feet 8 inches, then 35 feet 6 inches. You may not enjoy sensational improvement from this trick, but it should do no harm, and possibly overcome a problem of poor transmitter loading into the coaxial cable because of mismatch.

Over the Mountain. The unfortunate CB operator who lives at the bottom of a box canyon or is hemmed in by tall obstacles can leap-frog out of his dilemma by another matching trick. This is the case where you want to run a very long cable to an antenna which clears the obstructions—on an adjacent hill or other high formation. If the run is less than about 200 feet, the easiest way out is to install RG/8U cable. It's 50 ohms, but it operates at less loss per foot than the RG/58U style. But if you must run hundreds of feet, it would probably be cheaper to try another possibility. It's "open-wire" line; a ladder-like pair of conductors spaced about an inch apart by plastic insulators (the "rungs" of the ladder). This feeder type is sold for TV use on UHF channels because its loss is extremely low. Remember, though, the first limitation of open-wire feeders is that they must be insulated by standoffs all along the way (coax can run anywhere).

The other pitfall is impedance matching. These lines reduce loss by wide spacing and impedances of several hundred ohms. It means you can't connect the feeder directly to a CB rig or antenna since standard ratings are 50 ohms. Yet, if you must run open line to keep cost and losses down, you can transform impedance at either end and come up with a good match. It's done by "balun" coils that are sold for TV tuner repair.

The term "balun" derives from "balanced-unbalanced." These coils enable you to connect together unbalanced devices (CB rig and antenna) to a balanced device (the open-wire line). Secondly, the baluns provide about a four-to-one impedance transformation to allow these various devices to connect together with no mismatch. We experimented successfully with some inexpensive Saxton Model BC-2 baluns ordered from Lafayette Radio-Electronics. If you try these coils, use the following schematic, supplied in the coil package; "Circuit A—75 ohm to 300 ohm". One pair of coils connects the 75-ohm side to the CB transmitter; the second pair connects the 75 ohms to the antenna base. (Although 75 ohms is specified, the coils operate at very low SWR at CB's 50 ohms.) See Fig. 9.

It's Round Up Time. Finally, whatever matching technique or adjustment you follow in your CB rig, don't run the "ratio race". Although standing waves should have the lowest ratio indicated on an SWR meter inserted in the transmission line, anything below 2-to-1 is all you need for good signals on both transmitting and receiving sides. An overall look at the major matching points is shown in Fig. 10.

Antenna matching is a must for CBers. Admittedly you can transmit far beyond 150 miles on 5-watts input to your rig's final, but, far too many stations can't get past the next hill. ■

Oscillating Amplifiers

□ As any slightly cynical experimenter can tell you, if you want an oscillator, build an amplifier—it's sure to oscillate. Conversely, if you want an amplifier, (this same cynic will tell you), build an oscillator—it's sure to fail to oscillate, and you can then use it as an amplifier![1]

Our informed cynic must have had long and unhappy experience with negative-feedback amplifiers, which are known to have at least two outstanding characteristics:

1. They function beautifully if carefully designed and built.
2. Otherwise, they oscillate!

Why do they oscillate? Or, more basically, how does a feedback amplifier differ from an oscillator?

The fundamental block diagrams of an oscillator and an amplifier with feedback bear a strong resemblance to each other, as you can see from Fig. 1. From a block diagram viewpoint, both diagrams are *very* similar. Both contain some type of amplifying device, and both have part of their output signal fed back to their input. There are only two major differences between them:

1. The amplifier with feedback contains an *inverting* amplifier; the oscillator contains a *non*-inverting amplifier.
2. The oscillator doesn't have an input.

The circuit action obtained from these two circuits is entirely different. In the amplifier with feedback, the output waveform is upside down with respect to the input, so when it is fed back to the amplifier input, it cancels a portion of the input waveform. The output is therefore less than it would be without feedback. See Fig. 2.

The feedback signals from inventing amplifiers are not "in phase" with the input signal and subtract (or reduce) the input signal level to the amplifier. When a feedback signal does this, it is called *negative feedback*.

So Why Negative Feedback? Of course, if you merely want the biggest possible gain for your money, negative feedback's not your game. However, negative feedback of-

[1] This is well known as a corollary to Murphy's famous Law, "If anything *can* go wrong—it *will!*"

Fig. 1

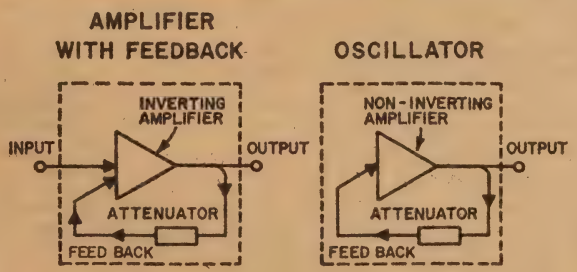
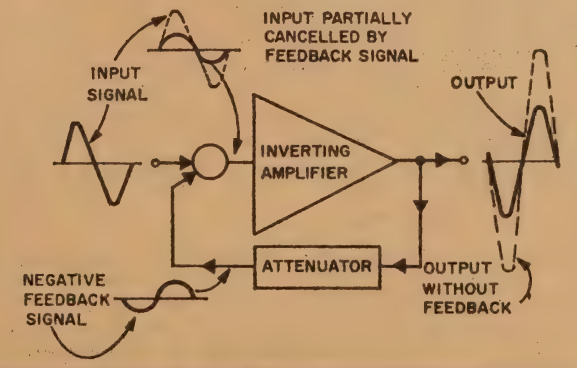


Fig. 2



fers other advantages, which can be summed up by saying that the amplifier's output, though smaller, is always nearly constant for the same input signal. For example, if the amplifier weakens with age, and the output tries to drop, there is less signal to be fed back; hence there is less cancellation, and the output is restored almost to its former level. Similarly, if you feed a high-frequency signal through the amplifier—so high in frequency that the amplifier can barely amplify it—the resultant drop in output reduces the fed-back voltage, produces almost no canceiling feedback signal and keeps the output nearly the same as it was at lower frequencies. Moreover, any clipping or other distortion of the waveform inside the amplifier produces an output waveform which does not match the input; hence the non-matching part is not cancelled, and the distortion is removed, or a least greatly reduced. Without this action, hi-fi amplifiers would not exist.

So the loss in output you obtain from negative feedback repays you by providing less distortion, better long-term stability, and better frequency response—that is, the best and most uniform output in response to *all* input frequencies.

On the Flip Side. The oscillator, on the other hand, is not supposed to give the best output from all input frequencies, but is instead made to give an output at a *single* frequency—with no input at all. It's not surprising that the opposite type of internal amplifier (non-inverting) is used to obtain this opposite result. See Fig. 3.

In the oscillator, any output at all (probably the result of some random noise in the internal amplifying device) is fed back, non-inverted, to the input, where it does not cancel but instead serves as the signal at the input. This feedback signal causes an even larger output, which results in an even larger signal fed back, further reinforcing the input signal, and so on.

You guessed it—this type of feedback signal is commonly referred to as *postive feedback*. In theory, the output waveform should continue to get larger forever. In practice, the amplifier is limited in the maximum size of the signal it can deliver, so the output waveform stops growing at this amplitude. As it stops to grow, so does the postive feedback signal. Now the signal reduces rapidly and the positive feedback signal lends a hand until the signal can get no lower. This is the beginning of the first cycle of many to follow.

All well and good, you say, but if the major difference between feedback amplifiers and oscillators is the inverting or non-inverting nature of their internal amplifiers, why does an amplifier sometimes oscillate? What turns an inverting amplifier into a non-inverting one?

To answer this question, first observe that an inverting amplifier, in passing a sine-wave signal, *effectively* shifts the signal's phase by 180° as shown in Fig. 4. We says *effectively*, because it doesn't really shift the timing by delaying the signal (which is what a real phase-shifter does) but, by turning the signal upside down, the amplifier makes it look like a signal which has been delayed (phase-shifted) by 180°.

A real phase-shifter, on the other hand,

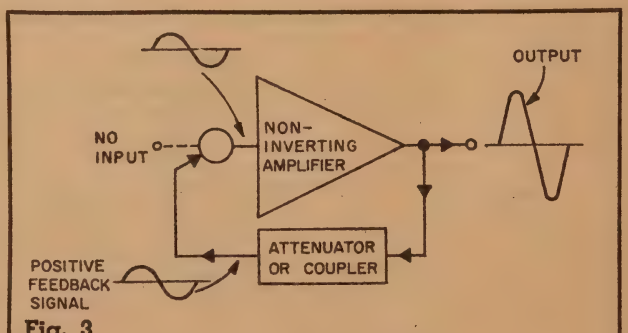


Fig. 3

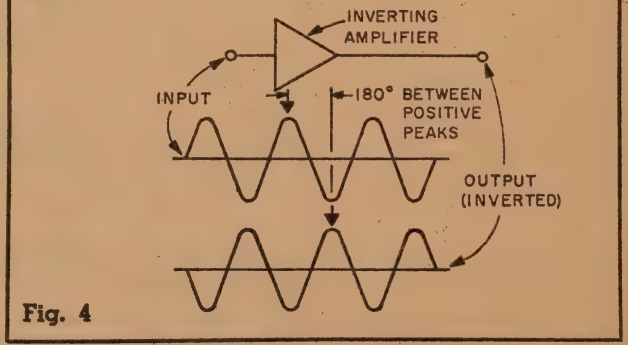


Fig. 4

is normally nothing but a fistful of judiciously connected resistors and capacitors (and sometimes inductors) which can be designed to give a 180° phase shift *at a single frequency*, such as 1,000 Hz, for example. In contrast to an inverting amplifier, it provides this phase shift by actually delaying the signal. See Fig. 5.

What happens if we combine an inverting amplifier and a 180° phase-shifter? Take a look at Fig. 6.

This combination will shift the phase of a given frequency by a total of 360° (an entire cycle) so the output is identical to the input. In effect, this combination (at 1,000 Hz) *will behave the same as a non-inverting amplifier*. See Fig. 6.

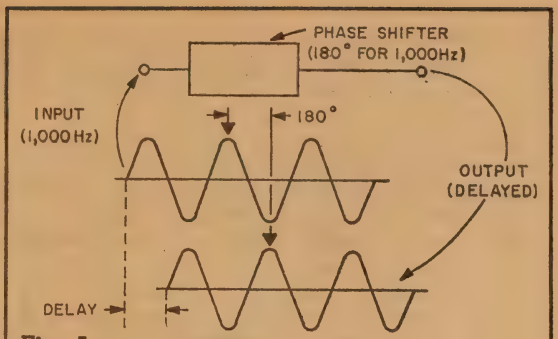


Fig. 5

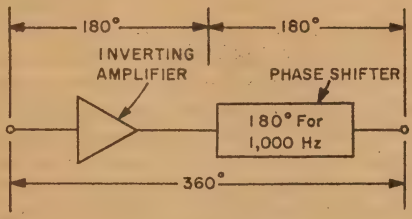


Fig. 6

Therefore, if we build a feedback amplifier which contains the normal inverting amplifier but also (inadvertently) contains a 180° phase-shift network, the resultant circuit will oscillate at the particular frequency, (1,000 Hz in the figure) for which the phase-shifter provides 180° phase shift. See Fig. 7.

How can one "inadvertently" make a phase-shifter? It's easier than you might think. The circuit shown in Fig. 8A will provide 60° phase shift at 1,000 Hz. Three such networks connected in a "ladder" (see Fig. 8B) will proved 3 X 60° = 180° of phase shift.[2] This network, if dropped into a normal feedback ampliflier circuit, will convert it to an oscillator.

[2] But not at 1,000 Hz. Because of the way the networks load each other, the 180° shift occurs at 707 Hz. However, if an amplifier were located between each network, then the amplifier will oscillate at 1,000 Hz.

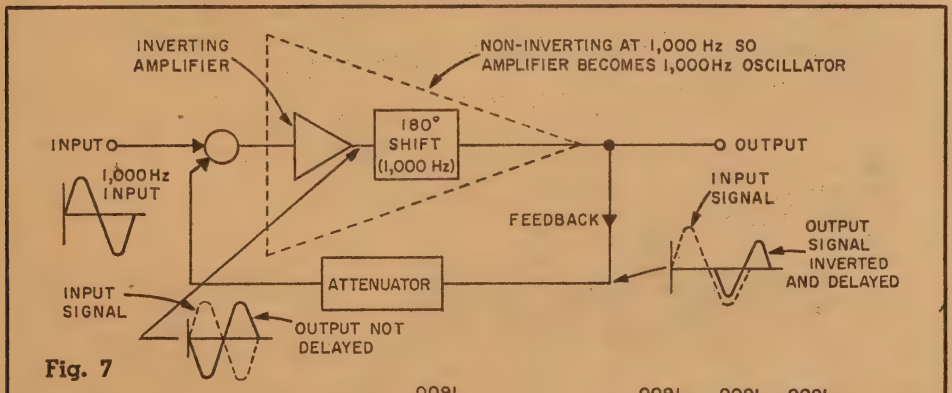


Fig. 7

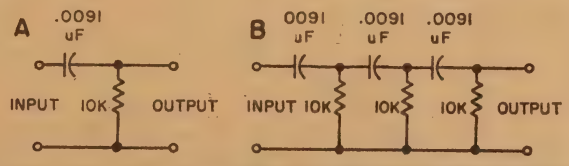


Fig. 8

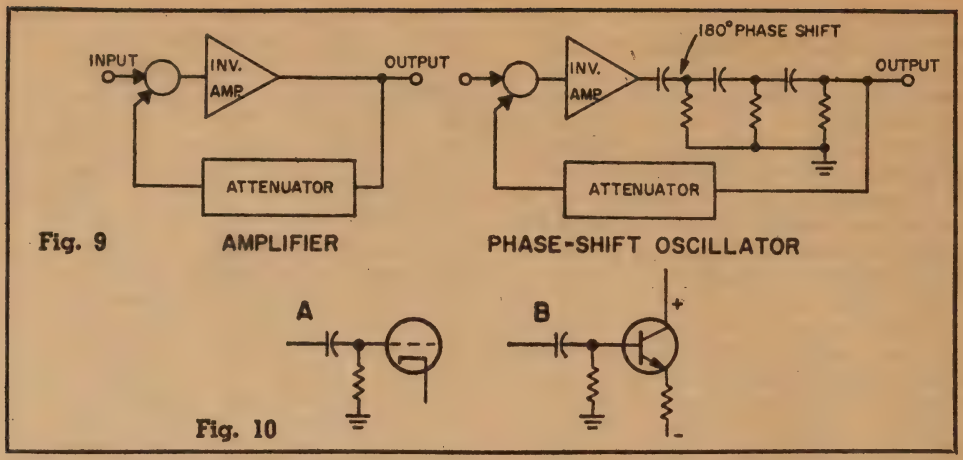


Fig. 9 AMPLIFIER PHASE-SHIFT OSCILLATOR

Fig. 10

This circuit (Fig. 9) is known as a *phase-shift oscillator* and is widely used in electronics.

Of course, when you set out to build a phase-shift oscillator, you *deliberately* insert a phase-shifter to make the circuit oscillate. How could one ever *inadvertently* place such a circuit in a feedback amplifier, thereby producing unwanted oscillations?

Phase-shift circuits can "hide" within an amplifier, posing as other circuits. For example, vacuum-tube amplifiers often have grid circuits arranged as shown in Fig. 10A. Does that resistor/capacitor circuit look familiar? In form, it's just like the phase-shift circuit above. And transistor amplifier circuits often take the form shown in Fig. 10B.

Again, the coupling/biasing network looks just like the basic phase-shifter network. At some frequency, this network will provide 60° of phase shift. If we use three such identical networks in a three-stage amplifier we have a 180° phase-shift network "buried" inside the amplifier, masquerading as three normal coupling networks. If this three-stage amplifier is used as part of a

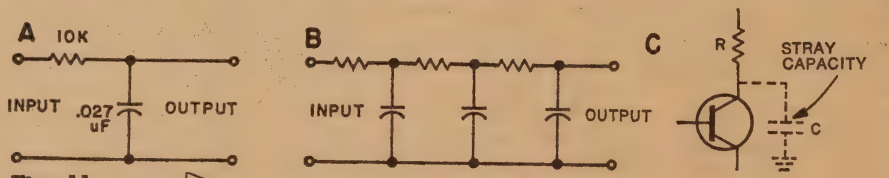


Fig. 11

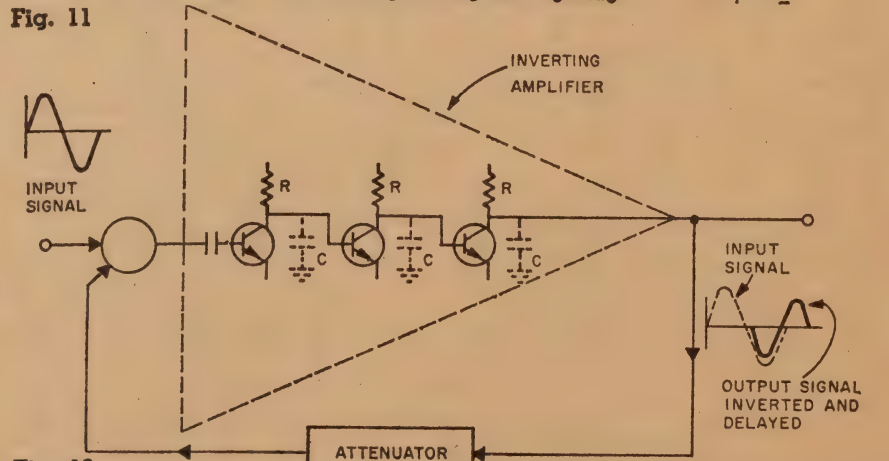


Fig. 12

feedback amplifier arrangement, the amplifier will oscillate at some frequency, and be quite useless for the purpose for which it was intended.

This is not the only way an amplifier can get into trouble. There are other types of phase-shifters than can creep into amplifiers, unrecognized, and drive the unwary experimenter up the nearest wall. This circuit (shown in Fig. 11A) can also produce a phase-shift of 60° at 1,000 Hz. Three of them, can produce the 180° phase-shift required for oscillation. See Fig. 11B. This particular network can invade amplifiers in an even more insidious fashion. The "masquerading" part of the circuit is shown heavy in Fig. 11C. The dotted capacitor doesn't appear physically in the circuit, because it is the so-called "stray" capacity associated with wires, sockets, terminals, etc. Three of these circuits hiding in an amplifier, can produce an unwanted oscillation. See Fig. 12. Since the stray capacities are so small, this "osc-plifier" will oscillate at a very high frequency; often so high that it is undetected as an oscillation. However, such oscillation can make an amplifier behave erratically; sometimes distorting, sometimes not; sometimes overheating, sometimes not. Fig. 7 and Fig. 12 have a lot in common.

Are feedback amplifiers the only culprits in this oscillating-amplifier business? Absolutely not! Often, so-called "straight" amplifiers—with no *intentional* feedback—will gaily oscillate away. But watch that word *intentional*. Close inspection of these misbehaving circuits usually uncovers an *unintentional* feedback path hiding within the amplifier. Consider the innocent-looking circuit in Fig. 13. This is an ordinary two-stage amplifier, obviously assigned the task of converting a small, positive-going signal into a large, positive-going signal. To help it along, the designer has even provided a decoupling network, R1 and C1. At high frequencies, C1 acts like a short circuit, effectively isolating (de-coupling) the amplifier's power bus, Ecc+, from the main power bus, Ecc++. But, at low frequencies, the capacitor acts like an open circuit—it just isn't there! A small part of the ouput voltage now appears across R1, and is coupled through the ampilfier's power bus back to

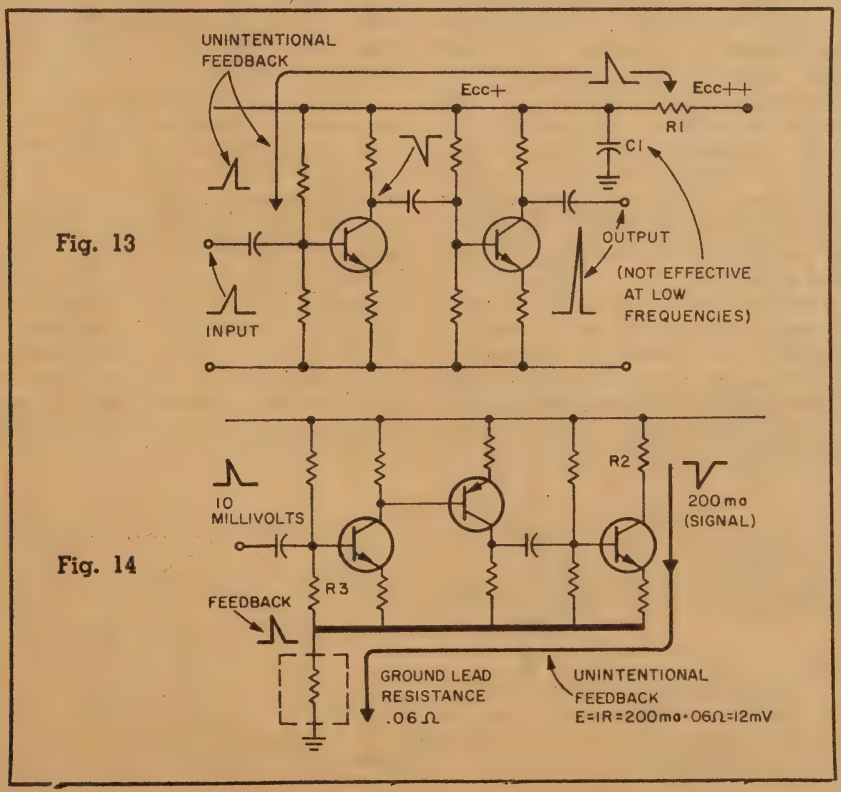


Fig. 13

Fig. 14

the input, arriving there with the same polarity as the normal input. True, the signal unintentionally fed back isn't very large, because the unintentional feedback path provides substantial losses for this stray signal. For example, the signal may arrive back at the input 100 times smaller than it was at the output. However, if the amplifier has a gain of 101, it makes an even larger output signal out of the fed-back signal, which then is fed back as an even larger voltage and oscillation begins.

Careless construction can get you into trouble, too. The amplifier shown in Fig. 14 is trying to convert a 10-millivolt input into a 200-ma signal needed by the load, R2. The builder has tied all ground returns to a heavy ground bus, and returned this bus to ground at only one point. Unfortunately, that single ground wire has to carry both the tiny input signal and the large output current. And, since every wire has *some* resistance, the actual circuit includes an 0.06-ohm resistor that does not appear in the original construction schematic diagram, but must be considered and is shown in Fig. 14.

Again, an uninvited, unintended feedback path has appeared, coupling the output back to the input. In the sketch, the large output current, flowing through the tiny ground-lead resistance, produces a voltage which is even larger than the original input voltage. And, since this voltage is also connected to the input (through the bias resistor R3), the fed-back voltage appears uninverted (and uninvited!) at the input, and will cause the amplifier to oscillate. What is to be done to convert these oscillators back into well-behaved amplifiers?

The general rule is *divide* and *conquer*. In the example just above, we can conquer the oscillation by dividing the ground returns, making sure that the high-current output circuits and the sensitive input circuits have their own private and individual paths to the power supply. See Fig. 15.

Short leads to the input connector are also helpful in squashing oscillations.

The misbehaving decoupling network, R1 and C2 in Fig. 13, can also be brought under control by dividing the network into *two* decoupling networks as shown in Fig. 16 below.

The stray capacities causing the unwelcome phase shifter to hide in the amplifier are harder to exorcise. Your best divide-and-conquer approach to this amplifier is to put feedback around only a pair of stages instead of three or more. This way, there are only two pairs of stray R's and C's lurking in the amplifier, and it takes at least three such pairs to make an oscillator.

The coupling capacitors, which combined to make a phase shifter in the very first example, can be prevented from ganging up on the amplifier and making it oscillate by making the product of each capacitor times its associated resistor (called the "RC Product") 5 or 10 times larger or smaller than the other RC products. For example, in a three-stage feedback amplifier which has all its base resistors the same values, you could make the three coupling capacitors 2μF, and 10μF, and 50μF, respectively. Again, you have divided the coupling capacitors into three widely-separated values, and conquered the oscillation.

All three problems and solutions are by no means a comprehensive list of all the ways an amplifier can get into trouble, nor of all the tricks of the trade that can be used to bail them out. For that matter, we've said nothing at all about the opposite problem—how to persuade a balky oscillator to "take off" and do its thing. But we've said enough to indicate that the cynic mouthing Murphy's Law would be well-advised to look to his basic theory and construction, because both amplifiers and oscillators answer to certain basic circuit laws, and will behave themselves impeccably if those laws are not violated. ■

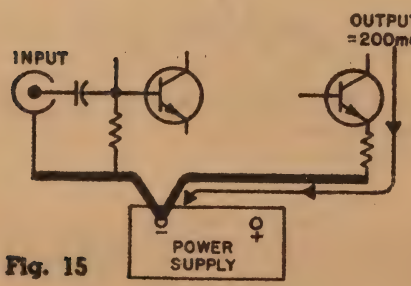


Fig. 15

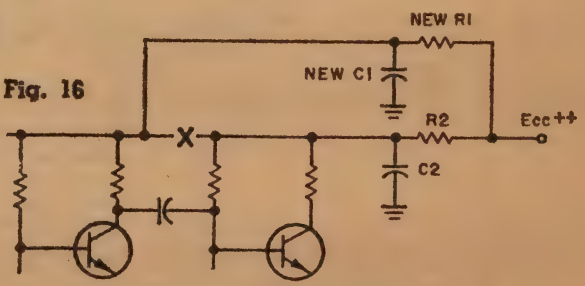


Fig. 16

Testing Batteries

☐ Each of us today uses a surprising number of batteries in daily living—more of them than we realize until we sit down to count all the battery-operated devices we have in home and shop. But how many of us have a good way to test all the various sizes and kinds of batteries we use, to learn when each should be replaced (or in some cases recharged)?

Mostly, we must rely on the old method of using a battery until it seems to be getting weak, and then replace it. If we want to be cautious, we may replace it sooner than necessary "just to be sure," and never really know if the old battery was really worn out.

Sometimes we may keep a couple extra batteries on hand for the most important devices in the household. These may go dead on the shelf before we get around to using them. So, the next time, we put off buying a replacement until we need one and then find the store is out of the size we need. Any way you look at it, we wind up with a feeling that "there must be a better way."

Naturally, There Is. It's through use of a battery tester that will check all the many types of batteries we have in the house. And to do the job right, it must do more than just measure the voltage—it must also put the correct *load* on the battery while measuring the voltage.

The little testers you find on the counter at an electronics parts store do this in a limited fashion. It's easy enough to build a similar tester to check just one or two types of batteries this way for yourself. But with so many different types of batteries in common use today, these simple testers still don't do the complete job. What you need is a battery tester with a built-in load which can be adjusted to suit any common type of battery which you are likely to have. The better battery tester illustrated here does just that, and yet is not much more complex than a simple tester which may work with only a few selected types of batteries.

Using commonly-available components and inexpensive, yet suitably accurate meters, this battery tester enables you to test just about any household or workshop battery you're likely to encounter, from the little button cells used in hearing aids and photographic exposure meters, up through medium-voltage radio "B" batteries.

A Terminal Voltage? This tester has both a voltmeter and a milliameter, and so it enables you to check the battery voltage while at the same time observing the amount of current being drawn from the battery. By adjusting the tester so it draws the amount of current for which the battery is rated by the manufacturer, you can quickly see if its terminal voltage drops significantly under this load.

Three rheostats, of different resistance values, are used in the tester to give you continuously-variable control of the load placed on the battery. This is the key to use of this better battery tester—it gives you the ability to easily impose the proper load on the battery so you can observe the effect this load has on the battery's terminal voltage.

Each of the tester's two meters has three ranges, selected with a rotary switch just below the meter. Voltage ranges are 0-3, 0-30, and 0-60 volts. Current ranges are 0.5, 0-50, and 0-500 mA.

Three rheostats give you a selection of load resistance from zero to 20,000-ohms. Only one rheostat is used at a time, with S1 used to select the rheostat in use. By covering the load resistance range with three rheostats whose resistance values overlap, this tester allows you to select a medium or low resistance setting with greater precision than if a single rheostat were used for the load.

Overrated Rheostats. When you look at the parts list, you may be surprised at the power ratings of these rheostats—they range from 12½ watts up to 50 watts. Why such a high power dissipation rating for the load resistors to be used on household batteries, which don't produce nearly that much power?

These high power ratings are needed to give you the flexibility to test a wide range of batteries. The power rating on a rheostat applies only to the full resistance of the rheostat. When you turn the slider to a lower resistance point, as you will do when using this tester, the power rating of the portion of the rheostat which is being used is reduced accordingly.

You certainly won't be sucking 50 watts of power out of the batteries you test, but you need rheostats heavy enough to safely dissipate lesser amounts of power when set for less than their full resistance.

In using the tester, you'll adjust a rheostat used as a load on the battery being tested until the milliameter shows you this load is drawing from the battery the amount of current for which it is rated. To determine this "load current," refer to the accompanying battery table, which covers many of the commonly-used batteries you're likely to encounter, or refer to a battery manual from the manufacturer of the batteries you use.

For example, the normal full rated load current for a 1.5-volt size "D" flashlight cell is shown as 150 milliamperes. To test such a cell on this better battery tester, follow this procedure.

1. Plug a pair of test prods into the two banana jacks at the lower left corner of the tester. Set the voltmeter range switch to "1x" (for the 0.3 volt range) and the milliameter range switch to "100x" (for the 0-500 mA. range).

2. Set the load range switch to "low." This

BATTERY TEST REFERENCE CHART

Maximum Current (mA)	Manufacturer Type Number					Type and Application
	RCA	Burgess	Eveready	Ray-O-Vac	NEDA	
1.5 Volts						
25	VS034A	Z	915	7R	15E	AA flashlight cell
80	VS035A	1	935	1C	14F	C flashlight cell
150	VS036	2	950	2D	13F	D flashlight cell
20	VS073	NE	904	716	910	Key chain lights & novelties
20	VS074	7	912	4DO	24F	AAA flashlight cell
25	VS334	930	1015	15	15	Transistor radios AA size
80	VS335	130	1035	14	14	Portable radios C size
150	VS336	2R	1050	2LP	13	Portable radios D size
25	VS734	920	815BP			Photoflash AA size
80	VS735	120	835BP			Photoflash C size
150	VS736	220	850			Photoflash D size
200	VS1074	AL-7	E92	MN-2400	24F	Alkaline AAA size
300	VS1334	AL-9	E91	MN-1500	15A	Alkaline AA size
500	VS1335	AL-1	E93	MN-1400	14A	Alkaline C size
500	VS1336	AL-2	E95	MN-1300	13A	Alkaline D size
4.5 Volts						
250	VS067	F3	736	A3	3	Portable "A"
6 Volts						
250	VS040C	F4M	509	941		Lantern service spring terminals
25	VS068	Z4	724	A2	2	Portable "A"
500	VS317	TW1	731	918	918	Lantern service binding posts
7.5 Volts						
70	VS065	C5	717	9	9	Portable radio "A"
50	VS129	B5	713	8	8	Portable radio "A"
80	VS315	D5	707	26	26	Portable radio "A"
9 Volts						
9	VS300A	P6	226	1600	1600	Transistor radios
150	VS301	D6PI	2506	1601	1601	Transistor radios
15	VS305	2N6	246	1602	1602	Transistor radios
30	VS306	D6	276	1603	1603	Transistor radios
20	VS322	M6	266	1605	1605	Transistor radios
8	VS323	2U6	216	1604	1604	Transistor radios
7	VS327	L6	206		1611	Transistor radios
80	VS330	C6X	2356N		1612	Transistor radios
12 Volts						
250	VS342	TW1S	732			Lantern service
13.5 Volts						
10	VS304	XX9	239	1900	1900	Transistor service
15 Volts						
1.5	VS704	Y10	504		220	Photoflash BC units
22.5 Volts						
2.5	VS084	U15	412	215	215	Photoflash BC units, transistor service
30 Volts						
2.5	VS085	U20	413	A210	210	Transistor radios
45 Volts						
40	VS014	A30	W359	206	206	Portable radio "B"
25	VS015	Z30	738	205	205	Portable radio "B"
4	VS086	U30	415	213	213	Portable radio "B"
50	VS112	5308	762S	709	709	Portable radio "B"
70	VS344	B30	484	207	207	Portable radio "B"

will select load resistor R3.

3. Touch the positive test prod to the positive battery terminal and the negative test prod to "low" rheostat control (R3) until the milliameter shows you are drawing 150 milliamperes from the battery, and as you do this, also watch the voltmeter. If it continues to indicate nearly 1.5 volts as the current flow is adjusted upwards until it reaches 150 milliamperes, the cell is in good condition, since you can see it is supplying its rated load current without a significant drop in voltage. However, if the voltmeter shows you the voltage dropped excessively as the current was adjusted upward to the value that's normal for this particular cell, then you know the cell is weak.

Incidentally, if you compare battery ratings published by different manufacturers of similar battery types, you may find a few variations in ratings. However, the versatility of this battery tester permits you to operate it to fit virtually any ratings to be found for low and medium voltage batteries.

Some batteries that have just a little life left will produce their rated load current for a short time before their terminal voltage begins to drop off. So, even if the cell tests good at first, hold the test prods in place for a minute or so and watch the voltmeter closely. If the voltage begins to drop off, you know the cell may be all right for short periods of operation, but that it is nearing the end of its useful life. Start thinking about replacements.

For Unknown Loads. In describing the test procedure for a size "D" flashlight cell, it was said to use the *low* resistance rheostat load. That's because the author, who designed this tester, knew from his calculations which load to use. Many times though, you may not be sure which rheostat to use, so just follow this simple rule. Start a test using the *high* rheostat (R1) set for its highest resistance, and work down from there. This is easy to do if you wire the unit so each rheostat will be set for maximum resistance when it is turned fully counter-clockwise as viewed from the front of the panel.)

In making a test, turn the resistance knob to the right, which will reduce the resistance, as you seek a setting that will produce on the milliameter an indication of the current for which the battery is rated.

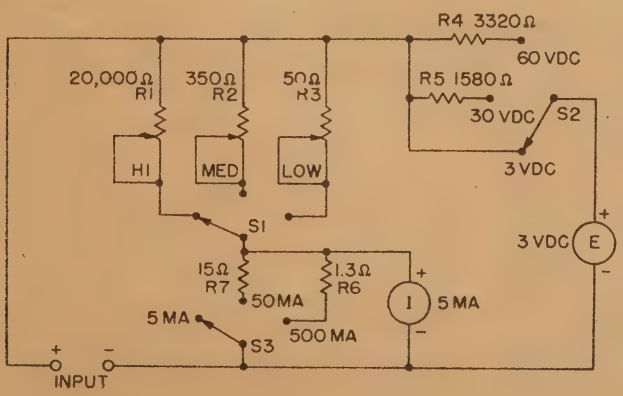
If you find you must turn the rheostat nearly fully clockwise to obtain the amount of current you want to draw from the battery being tested, you will find it's very hard to set the rheostat precisely enough to hold the milliameter steady. This will happen because you will be attempting to work with a very tiny portion of the rheostat's full resistance range.

When this happens, the thing to do is set the load selector switch (S1) to *medium* and try again, using the medium rheostat (R2). If the same thing occurs with R2, then set S1 to *low* and use the low rheostat (R3).

As a general rule, you'll find the high rheostat will be used on tests requiring fairly low amounts of current from low voltage batteries, and the low rheostat will be used for high amounts of current from low and medium voltage batteries. The medium rheostat will be used to conveniently bridge the gap between these two general categories. However, because of the wide variety of voltage and current combinations now available in the many battery types, it's not practical to set an iron-clad rule to determine which rheostat to use in every case. So, just follow the procedure of starting with a high resistance and working down until you reach the proper load for the battery you're testing.

Construction. Building the battery tester is very simple and straightforward, as you can see from the photos. Any type of metal or plastic utility box may be used. The one shown here is a 5 x 9½ x 2¼-in. steel converta-box, which is a little easier to work with than the more common minibox.

Drill ⅜-in. holes for the switches and rheostats. Drill large enough holes for the two binding posts to accommodate fiber shoulder washers so the binding post mounting screws will be insulated from the metal panel, and cut rectangular holes for the meters. ■



Parts You'll Need

E—0-3 volt DC meter, edge reading or round (Allied 701-5102 or equiv.)
I—0-5 mA DC meter, edge reading or round (Allied 701-5302 or equiv.)
R1—20,000-ohm, 50-watt rheostat (Ohmite 4211 or equiv.)
R2—350-ohm, 25-watt rheostat (Ohmite 0155 or equiv.)
R3—50-ohm, 12½-watt rheostat (Ohmite 0110 or equiv.)
R4—3320-ohm, ½-watt resistor (IRC type CEC-T-O or equiv.)
R5—1580-ohm, ½-watt resistor (IRC type CEC-T-O or equiv.)
R6—1.3-ohm, ½-watt resistor (Ohmite type RC20GF or equiv.)
R7—15-ohm, ½-watt resistor

Power Supply Basics

☐ The catalog of a leading electronics supplier contained this glowing description: A superhet shortwave receiver covering the standard broadcast band through 20 Meters. Its cabinet was luxurious walnut, its audio output push-pull into a high-quality speaker. The set boasted low current drain and the latest circuitry. The price?—a mere $49.75.

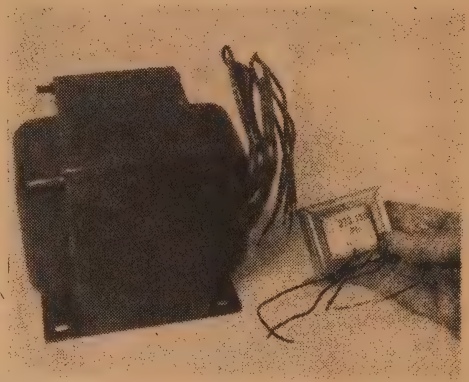
The catalog was Allied Radio's and the date was 1932. The radio was a console meant for the living room, and it no doubt pulled in the A&P Gypsies with reasonable fidelity. Thing is, it required batteries for power.

Here's the battery complement for the handsome, but hungry, *Knight 8* vintage receiver: Three 45-volt "B" batteries for tube plates; One 2-volt "A" cell for lighting filaments; One 22.5-volt "C" battery for biasing tube grids. This mountain of Eveready cost $9.00, a rather steep tab even in the good old days. And they could have pooped out right in the middle of a Herbert Hoover speech.

Super Supplies. That danger is gone, thanks to power supplies. Now a receiver takes a raw electricity from the utility company and converts it to filament, plate, or bias voltages. It does the same for transistorized circuits. Or it perhaps participates in the growing trend to 3-way operation, where you use the same device at home, in a car, or carry it as a portable. The supply not only powers the equipment in the home; it also recharges the poratble batteries. Cost is low because AC power is priced about 3¢ per kilowatt hour—which means you can operate a plugged-in table radio for about 100 hours on a penny.

Though power supplies operate circuits of vastly different voltage and current requirements, the basic principles are the same. In most instances a supply accepts house current—usually 117 volts AC alternating at 60 Hz (cycles)—and performs the following steps.

● *Transforming voltage.* The power company provides 117 volts for home outlets, but it's hardly the value that many electronic devices demand. The plates of receiving tubes require about 100 to 250 volts for operation, while transmitting tubes may need a "B+" several hundred volts higher. Transistors, on the other hand, usually function at least than 30 volts. So the first task of the supply is to transform voltage to the desired value. In many CB sets, for example, there's plate-voltage requirement of 250 and filament-voltage requirement of 12.6 VAC. The power transformer delivers these levels.



Typical power transformer sizes. Left, one for 1-kW ham rig; right, one for small receiver.

● *Changing AC to DC.* Furnishing correct voltage is not enough. Those voltages must often be DC—and the power company provides alternating current. So the second function of a supply is to *rectify*, or convert AC to DC. If a rectifier malfunctions in your radio you'll soon learn its function. The symptom is annoying hum in the speaker (caused by 60-Hz alternations in the audio). In a TV set, suffering rectifiers can put a thick, dark, "hum" bar across the screen.

● *Filtering.* Though rectifiers change AC to DC the product is far from suitable because it contains objectionable ripple. This will be attacked by the filter, which smooths the pulsations to pure DC.

The final step of the supply depends on the designer. He can add a *bleeder*, choose a *regulator*, or insert a *divider* at the output. We'll look at these extras, but first consider how the supply's basic parts operate.

The Transformer. In Fig. 1 is a typical power transformer that's been produced by the millions with only slight variations. As we'll see, the transformer acts to create a voltage change between its primary and various secondary windings. The trick's based on the turns-ratio between the various windings. If turns in the secondary number twice those of the primary, then output

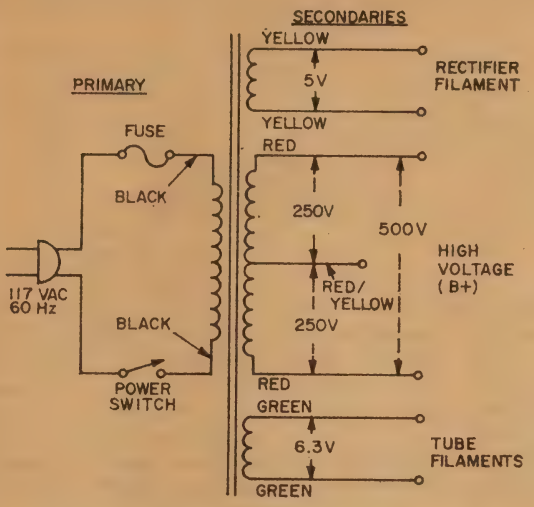


Fig. 1. Schematic of typical power transformer, showing voltages and EIA color coding of windings.

voltage doubles; if turns in the secondary are a fraction of those in the primary, then a stepdown in voltage occurs.

Thus, in Fig. 1, the rectifier filament, which operates at 5 volts, has few turns compared to the primary; the high-voltage winding at 500 volts, however, has about five times as many turns as the primary. The colors shown for the windings, incidentally, are standard and observed by many transformer manufacturers.

The *centertap* connection of a winding splits the voltage in half. In our example, the high-voltage secondary is capable of 500 volts across the full winding (red to red), but only 250 volts between the centertap (red/yellow) and either end. The most important job for a centertap occurs in a *full-wave* supply, as we'll see in a moment. Note that a protective fuse and a power switch are located in one primary lead of the transformer.

Rectification. The two filament voltages from our transformer (5.0 for the rectifier and 6.3 for other tubes) will need no further processing. AC can be applied directly for filament heating (or for lighting pilot lamps on the front panel). High voltage, however, must be converted to DC before powering tube plates or transistor collectors and drains.

A circuit for changing AC and DC is a *half-wave* rectifier, shown in Fig. 2. It's based on a diode's ability to conduct current in only one direction. The rectifier cathode boils off electrons (negative) which are attracted to the plate when the plate is driven positive by incoming AC.

When the next half-cycle of the AC appears, the plate is driven negative, so electrons are repelled at this time. The net result is shown in the output; a series of positive voltage pulses appearing at the load. (The dotted line shows where the negative side occurred.)

In practical circuits the half-wave rectifier is usually reserved for light-duty power supplies. It's inefficient because it fails to make

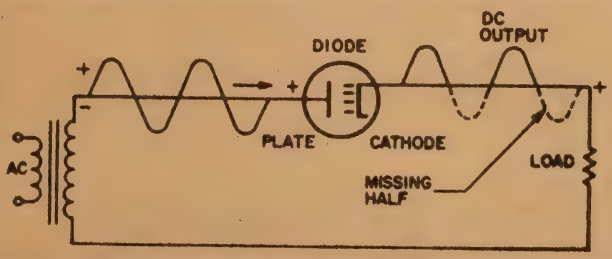


Fig. 2. Half-wave rectifier. showing how diode produces pulsating DC output by repelling electrons when plate is driven negative.

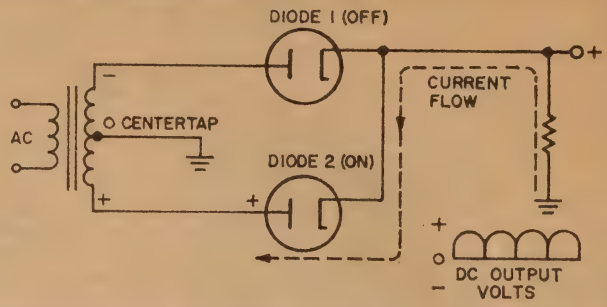


Fig. 3. Full-wave rectifier is pair of diodes driven alternatively to take advantage of both halves of AC input voltage.

use of AC voltage half the time (during the negative pulses). Secondly, those wide spaces between pulses are difficult to filter because of low ripple frequency. In a half-wave rectifier, the pulsations occur at 60 Hz, the same frequency as the applied line voltage. But don't underrate the half-wave supply because it's been used in just about every 4- or 5-tube table radio now playing. After all, its power requirements are low and the circuit is inexpensive to manufacture.

Full-wave Supplies. Transmitters and higher-power equipment overcome the half-wave's shortcomings with the full-wave system. It's nothing more than a pair of diodes that are driven alternately so they consume every bit of AC input voltage. The key to full-wave operation is the centertap on the transformer's secondary winding. As applied AC appears across the complete winding, it makes the top end negative (as shown in Fig. 3) and the bottom end positive.

The centertap at this time establishes the zero voltage point because it's at the common, or grounded, side of the circuit. During the time the lower diode (No. 2) has a positive plate, it does the conducting. Next, the applied AC voltage reverses and makes the top diode plate (No. 1) positive so this tube now conducts.

This load-sharing combination of two diodes and a centertapped power transformer not only improves efficiency, but doubles the ripple frequency. An input of 60 Hz emerges as 120 Hz in a full-wave arrangement because every half-cycle appears in the output. This reduces the pulsating effect (cycles are closer together) and the DC becomes easier to filter.

If you purchase a transformer, watch out for one pitfall. It may be rated, say, "250 volts CT" and appear to be suitable for a rig with a 250-volt plate supply. In a full-wave supply, however, the transformer voltage output would be only 125, since a centertap reduces the voltage of a winding by one-half. This can be avoided by specifying a transformer that has 250 volts *each* side of centertap or, stated another way, "500 volts CT."

Solid-State Rectifiers. Tube rectifiers are still widely found in electronic equipment, but they're destined for the Smithsonian Institute. Solid-state equivalents are superior because they don't need filaments or heaters to accomplish the same rectifying action. They're several hundred times smaller and much cooler in operation. Instead of a huge 5U4 vacuum-tube rectifier in your TV set, you're now more apt to find a pair of tiny silicon diodes.

Circuits using these semiconductors, though, are similar to those of vacuum tubes. As shown in Fig. 4, diodes can be used in equivalent half- and full-wave arrangements.

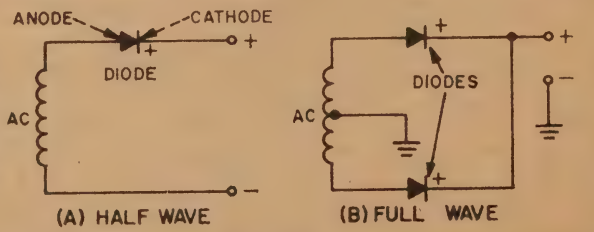


Fig. 4. Solid-state selenium or germanium diodes in similar half- and full-wave rectifier circuits are better than tubes since they don't need filaments or heaters for rectifying action.

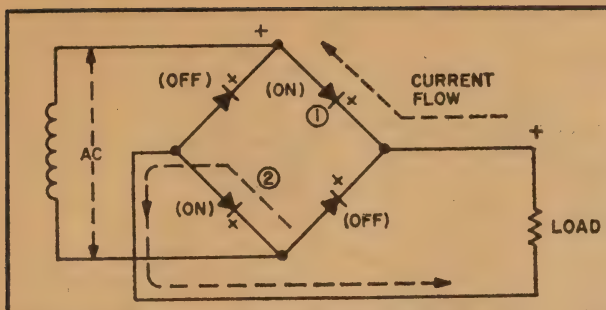
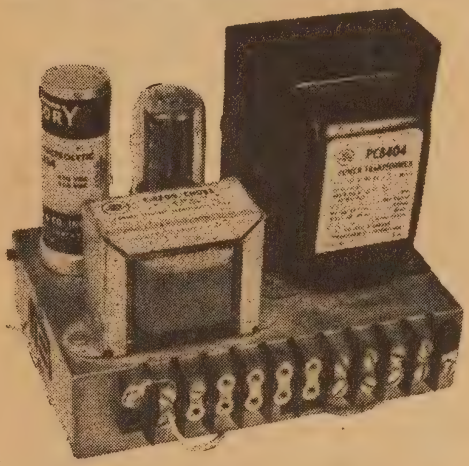


Fig. 5. Solid-state bridge rectifier provides full-wave rectification with center-tapped transformer by using four diodes.

Unlike tubes, though, solid-state diodes rectify AC and DC by a semiconductor effect at the diode junction (a region between the anode and cathode). The action, in simplified fashion, occurs when "current carriers" in the material flow toward and away from the junction under the influence of applied AC. When few carriers appear at the junction, little current gets through the diode; conversely, when many carriers are in the area, they reduce the junction's opposition to current flow. Depending on the way the diode is connected in the circuit, it can recover either the positive or negative half of the AC.

Bridge Rectifiers. Another common arrangement is the full-wave bridge (Fig. 5). Though it uses four diodes, it offsets this disadvantage by an ability to produce the same output as a regular full-wave supply without a centertapped transformer. It accomplishes the feat by operating one pair of diodes during each half cycle. And as one diode pulls current out of the load, its partner pushes current into it.

Net effect is a total voltage across the load which is about equal to the applied AC.



Filtering in this power-supply unit is by choke and capacitor seen at left of chassis.

We've shown how it occurs for diodes 1 and 2 in the diagram (Fig. 5) but a comparable action occurs in the other diodes when the AC switches polarity.

Filtering. Next major section of the

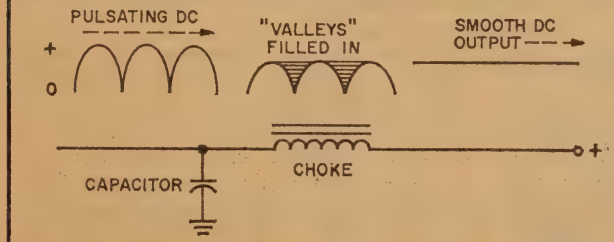
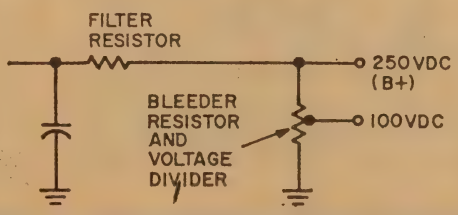


Fig. 6. Curve depicts how reactance of choke and discharge of capacitor eliminates ripple and fills in valleys for smooth DC.

Fig. 7. Choke can be replaced by filter resistor. Also shown is bleeder resistor that serves both as output regulator as well as voltage divider.



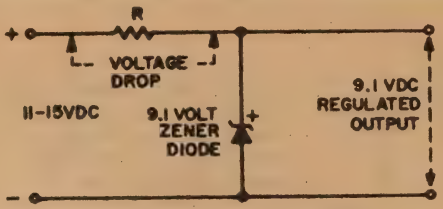


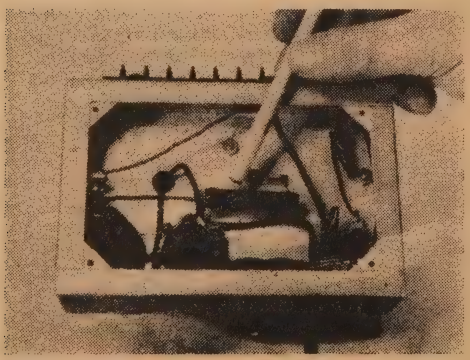
Fig. 8. For more efficient regulation than produced by bleeder resistor, Zener diode shunted across DC output will hold voltage to specified limit.

supply is the filter, which smooths out the ripple. Its two major components are often a *capacitor* and *choke* which eliminate pulsations by dumping a small amount of current from the peak of each ripple into the "valleys" between them. The result, as shown in Fig. 6, is pure DC fit for a tube or transistor.

In operation, pulsating DC arrives at the filter choke, a coil of wire wound on a soft iron core. As the name implies, the choke attempts to oppose any change in current flow. The rippling part of the wave, therefore, encounters high reactance in the choke and fails to get through. This is aided by the filter capacitor which is charged by ripple voltage.

As the ripple falls (between pulses), the capacitor discharges part of its stored current into the "valley." This the combined effect of choke and capacitor results in smooth DC which can have ripple as low as a few percent of the total voltage.

You won't find the choke in some power supplies because it's an expensive item. Many designers eliminate it (especially in mass-produced equipment) by using a resistor instead, as shown in Fig. 7. The resistor does the job of filtering but with one penalty; it reduces the amount of available voltage at the output. Yet, the loss can be tolerated in many circuits and filter resistors are common.



Bleeder resistor placed across rectifier/filter output acts as regulator.

Another use for resistors in a supply is to serve as a *bleeder*, also shown in Fig. 7. In this function, it protects parts in the supply from possible damage due to sudden voltage surges when the supply is first tuned *on*. Also, a bleeder helps stabilize voltage output when the load changes (as in a keyed ham transmitter) by always drawing some small degree of load current. Bleeders, too, are found in dangerous high-voltage circuits where they bleed off the stored charge of filter capacitors that could deliver a lethal shock to a repairman (even after the equipment has been turned *off*.)

Note that a tap can be added to the bleeder to provide a second output voltage from the supply. Now the bleeder becomes a *voltage divider*. As such, it can supply the designer with multiple output voltages for operating various devices in a circuit.

Voltage Regulation. A ham who's received a "pink ticket" from the FCC for chirpy signals, a color TV that's gone fuzzy, a shortwave receiver that won't stay on frequency—all may suffer from a problem in voltage regulation. Line-voltage fluctuations or other electrical swings can cause poor, unstable operation. So the engineers have come up with methods for "stiffening" a power supply.

If, say, line voltage changes from 105 to 130, they design the circuit to operate at 100 volts. Whatever voltage arrives over the line is reduced to 100, and the surplus is dumped (usually in the form of heat). To perform this task, the regulator establishes a reference point, then regulates around it.

A common example is the zener diode found in the power supply of many CB transceivers. Since these rigs can operate from a car's battery or generator, supply voltage can swing from 11 to 15 volts. This could happen if you're standing for a traffic light, then pull away, causing a shift between car battery and generator. If the CB set is on at this time, receiver tuning could

be thrown off because of large changes in local oscillator voltage.

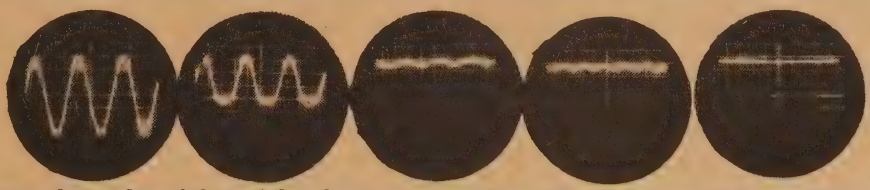
A zener diode can compensate for the shift, as shown in Fig. 8. At first glance it appears as an ordinary diode connected backward. Since the cathode (upper) terminal is connected to the positive side of the supply, there's a "reverse bias" condition. A zener diode, however, "breaks" down" (or "avalanches") whenever its rated (zener) voltage is exceeded. In our example, the zener is a 9.1-volt unit, so the diode conducts current as the supply voltage shifts from 11 to 15 VDC.

Yet we see 9.1 volts indicated at the output. Secret of the zener's ability to hold at 9.1 is that it detours part of the supply current as the voltage increases. Since a resistor is in series with that current flow, a voltage drop (as shown) appears across the resistor. Thus, any increase in supply voltage is dissipated across the resistor and effectively subtracted from the output. This automatic and continuous action occurs for any voltage above 9.1—the zener's nominal rating—so the output is said to be regulated.

More And Merrier. This barely brushes the subject of power supplies, since the variations are nearly endless. More than 20,000 volts for the picture tube of a color TV are derived from a special "flyback" transformer. It captures voltage from rapidly moving magnetic fields in the set's horizontal scanning section. An oscilloscope power supply contains strings of adjustable voltage dividers to move the pattern of light on the screen in any direction.

There are also high-current supplies with massive rectifiers for battery charging and super-smooth lab supplies for circuit design. But behind most of them are the simple principles which transform, rectify, filter, and regulate a voltage so it can do the job at hand. ■

If you want to replace your vacuum-tube rectifier with solid-state one, it's easy now with plug-in replacement shown here.



Scope photos, from left to right, shows (a) 60-Hz line voltage, (b) output from half-wave rectifier with negative portion eliminated, (c) ripple voltage from half-wave rectifier at same frequency as AC line voltage (d) frequency of ripple voltage from full-wave rectifier double that of AC line voltage (e) final DC output from power supply with no ripple voltage.

Facts on Frequency

☐ It was 4:00 a.m. and the thermometer read 15 degrees below zero. The squad car had to deliver the package without delay! Thanks to the flashing red light and the 2 kHz note screaming from the siren, the three miles from the bus terminal to police headquarters were covered in only 2½ minutes. The chief rushed out to get the package. He likes his coffee hot.

So the siren gave out a 2-kHz note. Is that anything like, say, the 60 Hz current which lights a table lamp or, maybe, the 27.155 MHz carrier a CBer sends out from his 5-watt rig? The answer is yes—and no!

No, because the siren note is actually a *disturbance of the air* which surrounds the whirling siren. The 60 Hz current which lights the lamp is actually a *disturbance of electrons* in the lamp cord. And the CB carrier is actually a *disturbance in the electromagnetic field* which surrounds the transmitting antenna. But the answer is also yes because, in spite of their apparent differences, the siren note, the 60 Hz current, and the 27.155 MHz signal all have something in common—the characteristic way each of these "disturbances" go through their vibrations.

Bouncing Air. If analyzed scientifically, each type of disturbance is seen to be a repetition of identical actions. When the siren gives out a constant howl, tiny air particles are forced to collide and spread apart—in steady rhythm. If the howl is a 2 KHz note, the colliding and spreading takes place 2000 times a second. Although the individual air particles move hardly any distance at all in their back and forth motion, the rhythmic collisions of these particles radiate from the siren in all directions, from particle to particle. This radiation is, of course, "sound" and travels through air at a speed of about 1100 feet per second.

Bouncing Electrons. Alternating current flows back and forth through a wire because electrons, in varying quantities, are made to push one another first in one direction and then the other—in steady rhythm. If this AC is 60 Hz current, such back and forth motion takes place 60 times a second. The individual electrons don't move very far along the wire in either direction but their "bumping" travels through the wire at a speed of almost 186,000 miles per second. Naturally, this bumping action is made to change directions in step with the electrons that cause it.

Bouncing Fields. Basically, a radio signal is a disturbance in which the electric and magnetic fields surrounding a transmitting antenna are distorted, first in one direction and then the other—in steady rhythm. If this signal is a 27.155 MHz CB carrier, these fields are forced to change direction 27,155,000 times a second. This rhythmic, field-reversing action radiates from the antenna at the speed of light, 186,000 miles per second.

Feel the Vibrations. Now—it's the changing nature of these three types of disturbances that we are really interested in! Aside from what is actually and physically happening, all three seem to follow the same pattern of change while going through their vibrations. True, the 27.155 MHz signal does its vibrating much, much faster than the other two but its vibration pattern is much the same.

Instead of using a lot of words to describe how each disturbance goes through its vibrations, let's use a helpful mathematical tool—the graph. The ancient Chinese said that *a picture is worth a thousand words* and that's exactly what a graph is and its worth. More important, a graph provides us with a *lasting* picture, a permanent record, of how these vibrations change their speed and direction.

Our graph is set up in the usual manner. First there are the two reference lines, the "vertical axis" and the "horizontal axis." In our graph, the horizontal axis is made to show the *passage of time*. How the vertical axis is used depends on which type of disturbance we are portraying. For the 2 kHz siren note, the vertical axis is made to measure *how far* each air particle moves during its back-and-forth motion. For the 60 Hz AC current, the vertical axis measures *how many* electrons are in mass movement along the wire during their forward-and-reverse motion. As for the 27.155 MHz signal, this axis tells *how much* and in what direction the electric and magnetic fields are being distorted.

So much for the preliminaries. Next comes the most important item. It doesn't matter which disturbance we are graphing—for each *instant of time* that is represented on the horizontal axis, there is a *point* above or below this axis which measures how far the vibration is displaced from its resting position. If all these individual points (and there are an unlimited number of them) are "plotted" on the graph, a continuous curve will appear. The amazing thing about this curve is that its shape looks pretty much the same for the 2 kHz note, the 60 Hz current and the CB carrier.

Study the special shape of this curve. While examining the curve (and this must be done from left to right, never from right to left!), notice that it rises abruptly, gradually tapers off, levels out for an instant, starts to fall, picks up downward speed, and falls below the horizontal axis only to repeat the process in upside-down fashion. The 2 kHz note will make 2000 of these waves in just one second while the 60 Hz curent goes through only 60 of them. But the

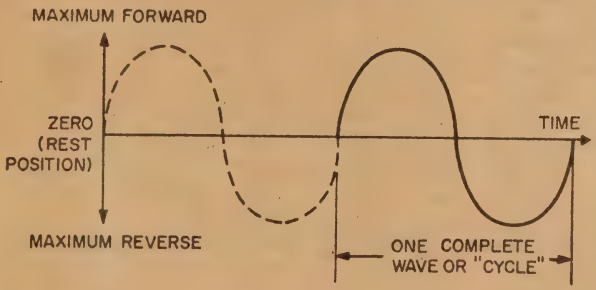
CB carrier will turn out 27,155,000 of these double swings during each second.

You Had Better Believe. This curve we have been discussing is something very special! It is fundamental to many different areas of science and engineering. It is the famous *sine curve.* The current leaving a simple AC generator varies according to a sine curve. So does the output of a vacuum-tube or transistor oscillator in an AM radio. Even the pendulum on a grandfather's clock swings back and forth in accordance with a sine-curve pattern.

But is must be understood and understood well! The sine curve is only a graph which shows the changing character of sound waves, alternating currents, and radio transmissions, etc. To be sure, a sine curve is a picture but it *is not* a picture of sound waves, currents and radio signals as they really look (if, in some way, we could actually see them).

Ironically (and students of electronics must not be misled because of this), there are sine-curve relationships in other fields of science where the physical occurance actually looks like a sine curve. If a pebble is dropped into a calm pond, for example, ever-enlarging circles (ripples) will be seen leaving the splash. However, if these ripples are viewed from the side, at water level, a train of honest-to-goodness sine curves will be observed as they leave the splash area. As another example of real-life sine curves, if two persons suspend a rope between each other and one of them whips it up and down, a beautiful sine wave will run along the rope. Nevertheless, sine curves *don't* travel through electrical circuits, they *don't* travel through air as sound waves, and they *don't* radiate from transmitting antennas! Radio signals, AC currents and sounds vibrate in unison with sine curves but they *don't* look like them! However, a check of their graphs indicate they do!

Discovering Frequency. At this point, the true meaning of "frequency" should become clear. It doesn't matter if we are talking about sound waves. AC currents or radio carriers. Frequency is the number of times these various disturbances repeat their vibrations during *one second.* From a graphical standpoint, frequency is the number of times *each* complete cycle of sine curve is repeated during *one second.* Either way you look at it, frequency can be classified by its number of "cycles per second." As everyone should now know, however, this once common designation for frequency has been replaced by, simply, "Hertz" (Hz) in honor of that great German scientist of the Nineteenth Century, Heinrick Hertz.



Whether you're trying to graphically picture ripples on a sylvan pond, or the unmodulated carrier of a Citizens Band RF voltage, the familiar sine wave is all you need to draw. All sine waves have peaks culminating in "maximum forward" points and "maximum negative" points, each occuring once each cycle.

To Sum Up. The 2000 Hz sound energy from the police-car siren, the 60 Hz AC from the wall outlet, and the 27,155,000 Hz carrier from the CB rig are three different scientific phenomena But they have the common property of being able to be described by sine curves and, because of this, all can be measured by a common yardstick —frequency.

But hold on! Are these three so-called disturbances really so different from each other? (After all, electronics wouldn't be electronics without sound and radio signals!) They are different but not as much as you would think. With the aid of a "transducer," one type can be *transformed* into another! Thus, a *microphone* will change 2 kHz sound waves into 2 kHz alternating current in a wire. A *loudspeaker* will change the AC back into sound waves. A *receiving antenna* will change 27.155 MHz electromagnetic energy into 27.155 MHz alternating current (AC) in a wire (the antenna feedline, that is). A *transmitting antenna* will make the opposite change.

That's it. If the interrelation between

sound, electrical and electromagnetic frequencies now makes sense to you, you've learned a tremendously important bit of electronics theory. And don't forget the almighty sine curve. The sine curve can be used to explain theory in many fields of science, not just electronics. Nevertheless, keep clear in your mind just how the sine curve fits into electronics—what it is and what it isn't. Maybe you don't care whether your coffee is hot but you better stay *hot* on the *sine curve*. ■

Multivibrators

□ When the conversation turns from the Mets to multivibrators, you may hear all manners of strange words bandied about. Flip-flop, one-shot, astable, and bistable roll off the tongue of the all-knowing. And don't be too surprised if you hear the words free-running, single-step, and monostable—to name just a few—at your next electrified cocktail party. What do all these terms mean? Are there really so many different kinds of multivibrators? And, for that matter, of what use is a multivibrator for the experimenter?

The main job of a multivibrator is to generate square waves and pulses. Period. That's all!

A square wave is often used as a test signal for audio amplifiers to reveal frequency-response problems. In other applications, multivibrators generate short time constant pulses—only a few microseconds in duration. These mini pulses synchronize, or steady, the picture on our TV screens.

Longer pulses—those which are several seconds in duration—control the exposure time of photographic enlargers. Slow multivibrators can also drive the flashing warning lights seen by motorists as they approach roadside hazards. And, in the radio amateur's shack, faster multivibrators running at audio rates train the ham's eye and ear as he works with his code practice oscillator. Or, the same MV, as the multivibrator's also called, doubles duty as an audio signal source. The list could go on and on.

The uses of multivibrators grow daily, limited only by the ingenuity of those who understand their working principles.

The imposing list of names in the first paragraph creates the impression that there must be almost a dozen different types of multivibrators. Fortunately, this is not so! There are only three basic types. The long list of names merely shows the existence of more than one name for the same type of multivibrator. At worst, this name-it hodge-podge reflects minor variations of the same type of multivibrator.

The Circuit With an Alias. The three basic types of multivibrators are the *free-running* multivibrator, the *one-shot* multivibrator, and the *flip-flop*. With these three basic circuit types under your belt, you can whip up any of the jobs a multivibrator's capable of doing.

The free-running multivibrator is probably the type most familiar to the experimenter. It is very likely that the square wave generator or oscilloscope on his workbench has a free-running multivibrator buried somewhere in the instrument's circuit. The outstanding characteristic of the free-running multivibrator—and the one from which it earns its name—is that it runs freely. As long as a power supply is connected to it, the free-running MV enthusiastically pumps out a never-ending stream of square waves. This feature consistently earns the title of the Most Popular Circuit whenever John Q. Electronicsbuff needs square waves. See Fig. 1.

In contrast to the free-running MV, the one-shot multivibrator is a very reluctant beast. If fed DC from a power supply, it does not react by joyously bubbling forth a stream of square waves like its enthusiastic free-running cousin. Instead, it sits there, grumpily doing absolutely nothing.

Fig. 1

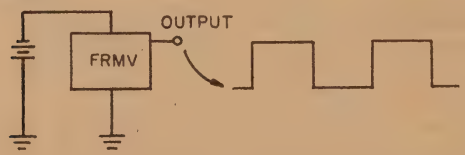


Fig. 2

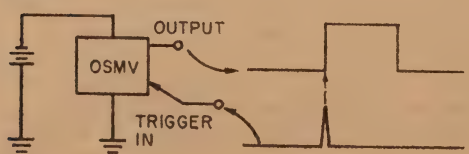


Fig. 3

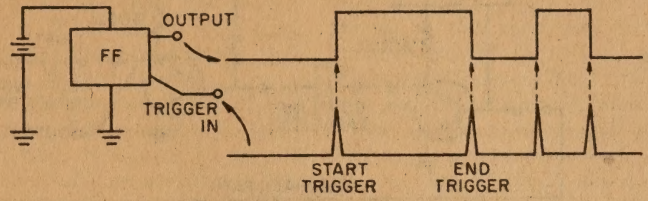
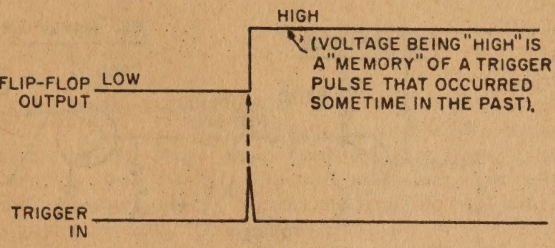


Fig. 4



And, it will continue to sit there unless kicked in the right place by an externally generated pulse, called a trigger pulse.

Under this urging, it reluctantly makes one and only one pulse, and then lapses back into its former sullen condition. Until, of course, it's kicked by another trigger. It derives its name—one shot—from the fact that it gives only one pulse in response to a trigger. See Fig. 2.

Flip Out Forget-me-not. The third type of multivibrator, the flip-flop, is a forgetful fellow. It, like the one-shot, gives no output pulse unless urged by a trigger pulse. But its response to a trigger is quite different. It starts out to produce a pulse, but forgets to end it, unless told to do so by another trigger pulse. Strangely enough, this forgetfulness can be turned into a memory. The flip-flop is the heart of the registers system of large computers. See Fig. 3.

How can this be? Because, as Fig. 4 shows, the flip-flop can remember forever (or, at least, until the power is turned off) that a trigger pulse has been applied to it. Using this single basic capability, the registers of giant computers can be constructed.

Some Basic Building Blocks. Circuit diagrams for these three basic multivibrators are surprisingly similar. They're all built from the same basic building blocks. These building blocks are shown in Fig. 5.

The free-running multivibrator combines these basic building blocks. The component values shown in our Fig. 6 will make a free-running multivibrator which runs at 440 Hz. Musicologists know that frequency as A above middle C on the piano.

To double that frequency, cut the values of both coupling capacitors in half; to triple it, cut them to one third the value shown, and so on. To hear the square wave, place an ordinary 2,000-ohm headset across either 1,500-ohm collector load resistor. To see the square wave, connect an oscilloscope to the point marked "output."

The one-shot multivibrator is very similar. It is built from pieces stolen from the

Fig. 5

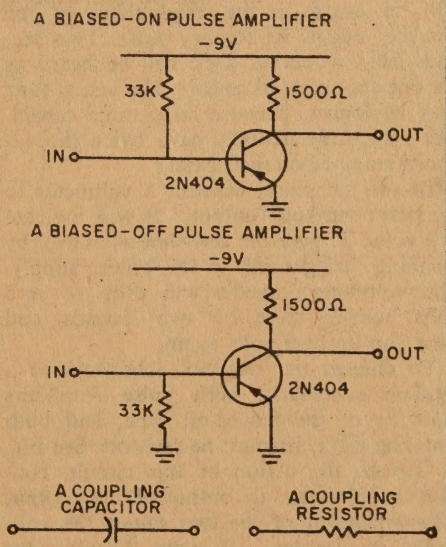


Fig. 6

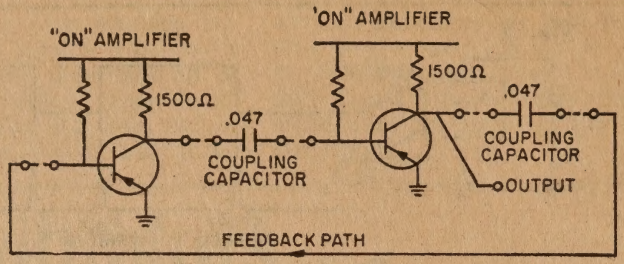
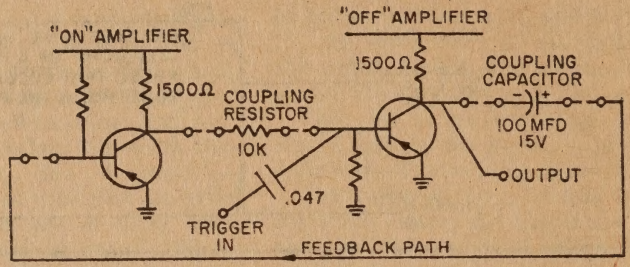


Fig. 7



free-running multivibrator as shown in Fig. 7 by replacing one of the coupling capacitors with a coupling resistor, and one of the "on" amplifiers with an "off" amplifier.

The values shown produce a pulse two seconds long. To double the pulse length, double the capacitor's value; to triple it, triple the capacitor's value, and so on. To hear the pulse, place an ordinary 2,000-ohm headset across either 1,500-ohm resistor.

Momentarily touch the point marked "trigger in" to the power supply. A click will be heard in the headphones as the one-shot begins its solitary pulse. Two seconds later, a second click will be heard as the one-shot ends its pulse. (The actual time may be longer, because large-value capacitors sometimes actually have twice the capacity stamped on their case.)

To see the pulse, connect a voltmeter to the point marked "output." It will indicate −9 volts. Trigger the one-shot as above, by touching "trigger in" to the power supply. The voltmeter's needle will drop to zero volts, remain there for two seconds, and then pop up to 9 volts again.

To change the one-shot schematic to a flip-flop schematic, both pulse amplifiers must be of the biased-off type, and both coupling elements must be resistors. See Fig. 8. To see the action of this circuit, connect a voltmeter to output #1 or output #2—whichever of the two causes the voltmeter to register −9 volts. Leaving the voltmeter connected, short the output to ground. The voltmeter reading will drop to zero, of course, because there is a dead short right across its terminals. But, the surprising thing is that the reading will stay at zero after the short is removed.

Next, short the other output to ground. The voltmeter reading will rise to −9 volts, and stay there after the short is removed, showing that the flip-flop can remember an occurrence (like shorting one output) even after the occurrence is ended.

Kicked by a Trigger Pulse. Of course, shorting an output to stimulate the flip-flop into action is not the same as running it from a trigger. Triggering circuitry can be added to the basic flip-flop as shown in Fig. 9. Now, leaving the voltmeter connected to one output as above, you can trigger the flip-flop by momentarily connecting the point marked "trigger in" to ground. Each time a trigger is supplied, the output that was at zero volts will jump to −9 volts. A second trigger will cause the same output to revert to zero volts.

Since it takes two triggers to make one complete output pulse from a flip-flop, feeding 500 pulses per second to the trigger input will cause only 250 pulses per second to come from the output. If these pulses are, in turn, fed to another flip-flop, its output will provide only 125 pulses per second. This ability of flip-flops to act as a frequency divider find very wide usage in

applications ranging from computers to TV and electronic organs.

So much, then, for the three basic types of multivibrators. What about that long list of names we bandied about in the first paragraph? Where do they come in?

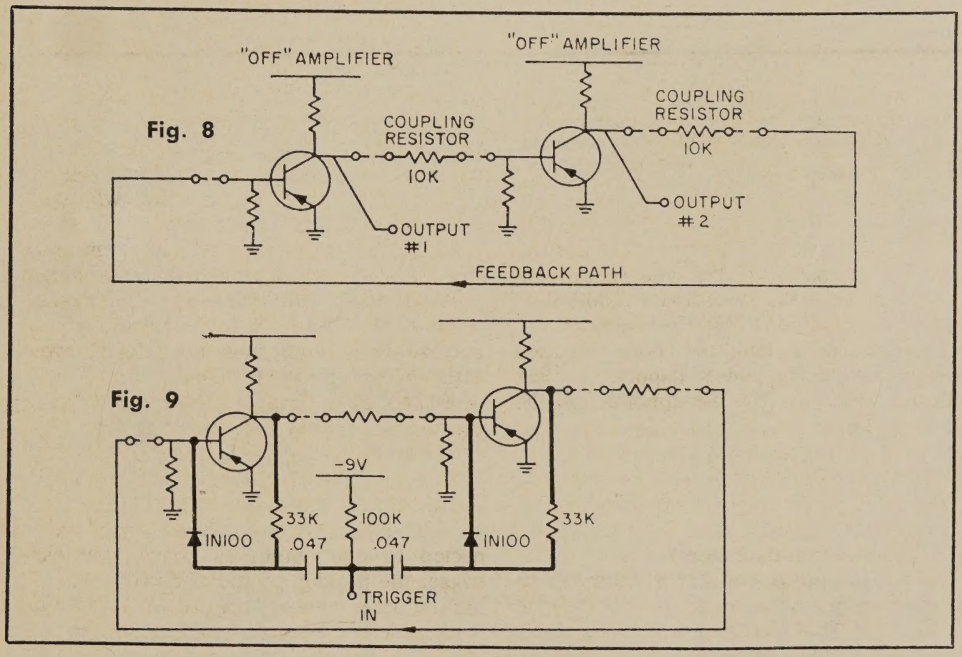
We can parcel out all those names among the three basic types. For example, the free-running multivibrator is known in formal electron-ese as an *astable* multivibrator. The prefix "a-" tells us that it lacks a stable condition, and hence runs endlessly as long as power is supplied. The one-shot MV also answers to the name *monostable* multivibrators, its one "mono" stable state being the one in which it grumpily sits while awaiting a trigger.

And, the flip-flop is also a *bistable* multivibrator because it has two, or, "bi-" stable states, and will sit happily forever with a given output either "high" (−9 volts in the above example) or "low" (zero volts).

Other Names, Yet. Other names for the flip-flop include *toggle*, from its action in response to two successive triggers; *binary*, from its ability to rest in either of two states, and *Eccles-Jordan*, after the two men who described the circuit many years ago.

The names *one-step* and *univibrator*, as well as *single-step*, *single-cycle*, and *monovibrator*, are all much less common names for the one-shot. Similarly, the free-running multivibrator is rather infrequently referred to as an *unstable* multivibrator or an *Abraham-Bloch circuit*.

So, in spite of the abundance of names, there are only three basic types of multivibrators. Call the MV what you will, but the application of these three types reach through to almost every project the electronics hobbyist is likely to conjure up on his workbench. ■



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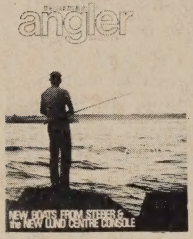

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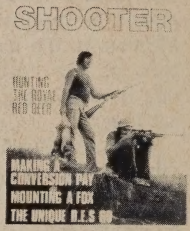

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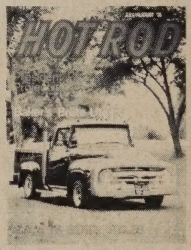

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